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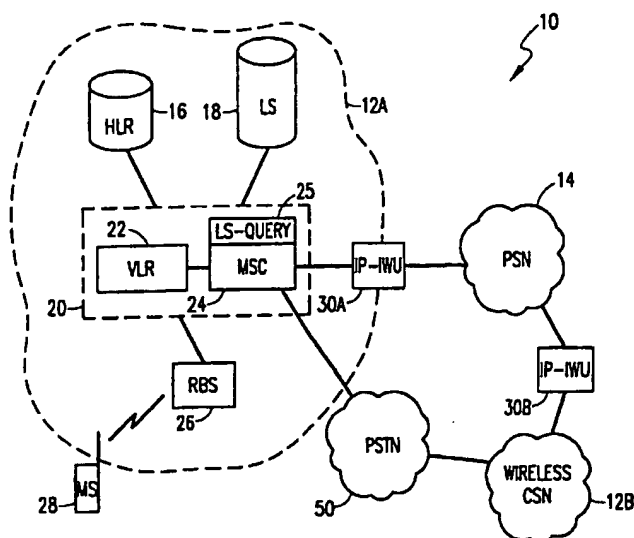
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(54) Title: SYSTEM AND METHOD FOR CALL ROUTING IN AN INTEGRATED TELECOMMUNICATIONS NETWORK HAVING A PACKET-SWITCHED NETWORK PORTION AND A CIRCUIT-SWITCHED NETWORK PORTION



(57) Abstract: A system and method for IP-based call routing in an integrated telecommunications network having a packet-switched network portion (e.g., a Voice-over-Internet Protocol (VoIP) network portion (14) and one or more circuit-switched network (CSN) portions (12) such as a PSTN (50) or a radio telephony network. A mobile Switching Center (MSC (24)) serving one or more mobile subscribers (28) is provided with an Internet Protocol (IP)-Interworking Unit interface (30) towards the VoIP network portion. The radio telephony network also includes a Location Server (LS(18)) containing mapping information between routing numbers (e.g., Temporary Location Directory Numbers or TLDNs), called party numbers (B-numbers) and IP addresses of entities to which a call can be routed over an IP trunk from the MSC. A querying mechanism is provided in the MSC for interrogating the LS based upon a routing number, a called party number, or both, provided to the MSC. The MSC obtains an IP address from the

LS which is used for effectuating the IP trunk. A plurality of Bearer Independent Call Control (BICC) messages and a plurality of Integrated Services Digital Network (ISDN) User Part (ISUP) messages are transmitted among the various nodes of the integrated telecommunications network, e.g., one or more MSCs with the IP interface, a Local Exchange of the PSTN, etc. for establishing the IP trunk. Where an IP trunk is not available, a circuit-switched path (e.g., a Synchronous Transfer Mode (STM) trunk), is utilized in the call path. The IP trunk is implemented using Real-time Transfer Protocol (RTP) and Session Description Protocol (SDP) to convey the voice payload associated with the call.

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**SYSTEM AND METHOD FOR CALL ROUTING IN AN INTEGRATED  
TELECOMMUNICATIONS NETWORK HAVING A PACKET-  
SWITCHED NETWORK PORTION AND A CIRCUIT-SWITCHED  
NETWORK PORTION**

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**BACKGROUND OF THE INVENTION**

Technical Field of the Invention

The present invention relates to integrated telecommunication systems and, more particularly, to a system and method for routing long-distance calls in an integrated telecommunications network having a packet-switched network portion (for example, a network using Internet Protocol (IP)) that is coupled to circuit-switched network portions such as a wireless telephony network portion, a Public Switched Telephone Network (PSTN), or both.

Description of Related Art

Coupled with the phenomenal growth in popularity of the Internet, there has been a tremendous interest in using packet-switched network (PSN) infrastructures (e.g., those based on IP addressing) as a replacement for the existing circuit-switched network (CSN) infrastructures used in today's telephony. From the network operators' perspective, the inherent traffic aggregation in packet-switched infrastructures allows for a reduction in the cost of transmission and the infrastructure cost per end-user. Ultimately, such cost reductions enable the network operators to pass on the concomitant cost savings to the end-users. One of the well-known advantages of the IP-based networks with respect to voice transmission is that considerable savings in long distance charges may be realized for calls that are to be routed over multiple geographic regions such as, for example, Local Access and Transport Areas (LATAs).

Some of the market drivers that impel the existing Voice-over-IP (VoIP) technology are: improvements in the quality of IP telephony; the Internet phenomenon; emergence of standards; cost-effective price-points for advanced services via media-rich call management, et cetera. One of the emerging standards in this area is the well-known H.323 protocol, developed by the International Telecommunications Union

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(ITU) for multimedia communications over packet-based networks. Using the H.323 standard, devices such as personal computers can inter-operate seamlessly in a vast inter-network, sharing a mixture of audio, video, and data across all forms of packet-based network portions.

5           The H.323 standard defines four major types of components for forming an inter-operable network: terminals, gateways, gatekeepers and Multipoint Control Units (MCUs). In general, terminals, gateways and MCUs of an H.323-based network are referred to as "endpoints." Gateways are typically provided between networks (or network portions) that operate based on different standards or protocols. For example,  
10       one or more gateways may be provided between a packet-switched network portion and a circuit-switched network portion. Terminals are employed by end-users for accessing the network or portions thereof, for example, for placing or receiving a call, or for accessing multimedia content at a remote site.

          The gatekeeper is typically defined as the entity on the network that provides  
15       address translation and controls access to the network for other H.323 components. Usually, a gatekeeper is provided with the address translation capability for a specified portion of the network called a "zone." Typically, a zone comprises all terminals, gateways, and MCUs (that is, all endpoints) managed by a single gatekeeper. Accordingly, a plurality of gatekeepers (sometimes referred to as a "gatekeeper cloud")  
20       may be provided for managing the entire network, each gatekeeper being responsible for a particular zone. In addition to address translation, gatekeepers may also provide other services to the terminals, gateways, and MCUs such as bandwidth management and gateway location.

          Those of ordinary skill in the art should appreciate that although the current  
25       VoIP networks offer rudimentary location services, they are not adequate for the mobility management required of a wireless network. In part, this deficiency is due to the condition that the gatekeeper which provides for call routing services and the registration of other H.323 entities within the VoIP network is typically unaware of conventional telecommunications terminals. While this condition is not a problem for  
30       fixed wireline telephones in terms of providing savings in long distance charges (one of the most important economic motivations behind IP-based call routing), calls

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involving mobile stations (MSs) may still require establishing long distance circuit-switched trunks from one Mobile Switching Center (MSC) to another for routing. For example, when a call to a mobile subscriber is received in a gateway MSC (GMSC), it queries a Home Location Register (HLR) of the mobile subscriber for the location of the serving MSC. The HLR, in turn, queries the serving MSC (or, visited MSC or VMSC) for a Temporary Location Directory Number (TLDN) for routing the call to the mobile subscriber. The TLDN is then passed to the GMSC for routing the call using circuit-switched trunks. Since the mobile station is no longer present in its home area, the GMSC-VMSC call leg can be a long distance call between two neighboring regions such as LATAs, two LATAs geographically separated from each other, or across a continent. Clearly, routing such long distance call segments over CSN portions defeats the rationale behind the use of VoIP network portions in integrated telecommunications networks having CSN portions.

It should also be understood that there are situations in some integrated telecommunications networks where long distance call segments are used even when the mobile subscriber is not roaming. For example, when the MSC that services the call originating mobile station (MS) and the MSC serving the home area of the terminating MS (that is, home gateway MSC) are situated in two different regions (e.g., LATAs), and the terminating MS is located in its home area, the call path still involves a CSN-based inter-MSC long distance trunk. Again, the economic advantages of a VoIP network are not achieved in such situations. Similarly, calls between a mobile station and a wireline phone served by a Local Exchange (LE) of the PSTN are also typically routed over CSN-based long distance trunks if different geographic regions are involved. Providing for IP-based call routing in such situations also gives rise to savings in long distance charges.

Based on the foregoing, it can be readily appreciated that there is an acute need for a solution that provides IP-based call routing so that the benefits of integrating VoIP network portions with CSN portions in an integrated telecommunications network are realized. The present invention provides such a solution.

## SUMMARY OF THE INVENTION

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In one aspect, the present invention is directed to an integrated telecommunications network having a packet-switched network portion (e.g., a Voice-over-Internet Protocol (VoIP) network portion) and one or more circuit-switched network (CSN) portions such as a PSTN or a radio telephony network. A Mobile Switching Center (MSC) serving one or more mobile subscribers is provided with an Internet Protocol (IP)-Interworking Unit interface towards the VoIP network portion. The radio telephony network also includes a Location Server (LS) containing mapping information between routing numbers (e.g., Temporary Location Directory Numbers or TLDNs), called party numbers (B-numbers) and IP addresses of entities to which a call can be routed over an IP trunk from the MSC. A querying mechanism is provided in the MSC for interrogating the LS based upon a routing number, a called party number, or both, provided to the MSC. The MSC obtains an IP address from the LS which is used for effectuating the IP trunk. A plurality of Bearer Independent Call Control (BICC) messages and a plurality of Integrated Services Digital Network (ISDN) User Part (ISUP) messages are transmitted among the various nodes of the integrated telecommunications network, e.g., one or more MSCs with the IP interfaces, a Local Exchange of the PSTN, etc. for establishing the IP trunk. Where an IP trunk segment is not available or possible, a circuit-switched path such as, e.g., a Synchronous Transfer Mode (STM) trunk, is used for completing the call routing path.

In another aspect, the present invention is directed to several embodiments of an IP-based long distance call routing method. In one embodiment, the call routing method relates to routing a call originated by a PSTN phone to an MS disposed in the integrated telecommunications network comprising the infrastructure as set forth above. In another embodiment, the call routing method relates to routing a call from an MS to a PSTN phone served by a Local Exchange. In yet another embodiment, the call routing method of the present invention is directed to routing a call originated by an MS to another MS that is located in its home area. In a still further embodiment, the call routing method relates to routing an MS-originated call to an MS that is located outside its home area.

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## BRIEF DESCRIPTION OF THE DRAWINGS

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A more complete understanding of the present invention may be had by reference to the following Detailed Description when taken in conjunction with the accompanying drawings wherein:

FIG. 1A depicts a functional block diagram of an integrated telecommunications network provided in accordance with the teachings of the present invention;

FIGS. 1B and 1C depict two scenarios, respectively, of a routing scheme for MS-to-MS calls wherein the called MS is located in a home system of an integrated telecommunications network;

FIGS. 2A and 2B depict a flow chart of a call routing method for MS-to-MS calls wherein the called MS is located in a home system;

FIGS. 3A and 3B depict two scenarios, respectively, of a routing scheme for MS-to-MS calls wherein the called MS is roaming in a visited system of an integrated telecommunications network;

FIGS. 4A - 4D depict a flow chart of a call routing method for MS-to-MS calls wherein the called MS is roaming;

FIGS. 5A and 5B depict two scenarios, respectively, of a routing scheme for PSTN-to-MS calls in an integrated telecommunications network;

FIGS. 6A - 6C depict a flow chart of a call routing method for PSTN-to-MS calls in an integrated telecommunications network;

FIGS. 7A and 7B depict two scenarios, respectively, of a routing scheme for MS-to-PSTN calls in an integrated telecommunications network; and

FIGS. 8A - 8C depict a flow chart of a call routing method for MS-to-PSTN calls in an integrated telecommunications network.

## DETAILED DESCRIPTION OF EMBODIMENTS

In the drawings, like or similar elements are designated with identical reference numerals throughout the several views, and the various elements depicted are not necessarily drawn to scale. Referring now to FIG. 1A, depicted therein is a functional block diagram of an integrated telecommunications network 10 provided in accordance with the teachings of the present invention. It should be appreciated that

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the integrated telecommunications network 10 is provided herein in order to exemplify the network-level infrastructure used in the various call routing scenarios described in greater detail hereinbelow.

5 The integrated telecommunications network 10 comprises a PSN portion 14 such as, for example, an H.323-based Voice-over-IP (VoIP) portion, that is coupled to a plurality CSN portions including, for example, one or more wireless telephony network portions (e.g., WL-CSN portions 12A and 12B) and a PSTN portion 50.

10 It should be readily apparent that the wireless CSN portions of the integrated telecommunications network 10 may be realized in any known radio telephony technology, for example, a Time Division Multiple Access (TDMA), et cetera. The WL-CSN portion 12A is shown in greater detail. A Home Location Register (HLR) 16 is provided for maintaining a subscriber profile or record associated with a mobile subscriber / mobile station (MS) 28. A Radio Base Station (RBS) 26 is included as part of the cellular infrastructure that comprises the WL-CSN portion 12A, in order to provide radio access services to the MS 28. A serving system 20, comprising a Visitor Location Register (VLR) 22 and a Mobile Switching Center (MSC) 24, is also included as a switching node therewith.

20 In accordance with the teachings of the present invention, an IP-Interworking Unit (IWU) is provided as a hardware/firmware platform for interfacing and interworking between the switching node (i.e., the MSC/VLR combination in this exemplary embodiment) of the WL-CSN portion 12A and the PSN portion 14. Preferably, the IP-IWU 30A is provided as an IP interface to the MSC 24, and includes appropriate media gateway (MGW) functionality for carrying voice traffic (i.e., payload) over the IP-based PSN portion 14.

25 In addition, a Location Server (LS) 18 is provided as an entity within the WL-CSN portion 12A that operates as a query-able database containing, preferably, mappings between CSN-based routable numbers (e.g., a called party's number (i.e., the B-number) or a Temporary Location Directory Number or TLDN) and an IP-network address of a signaling endpoint (e.g., an MSC having the IP-IWU interface)). Preferably, the switching node (e.g., the MSC/VLR combination 24/22 or the MSC 24 separately) includes a hardware/software/firmware-based LS-query function 25 that

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facilitates interrogation by the MSC of the LS 18. Further, the LS is preferably configured in such a way that it returns a unique IP-network address of a "destination MSC" which, in some instances, may be an MSC that serves a called MS. On the other hand, if the MSC associated with the called MS does not have an IP interface  
5 (i.e., not IP-addressable), the LS is configured so as to return the IP address of an MSC that is located closest thereto. With respect to PSTN calls, the destination MSC is the terminating IP signaling point connected to a Local Exchange (LE) disposed in the PSTN 50.

Based on the foregoing, it should be readily appreciated that the provision of  
10 a database as set forth above in conjunction with a query function provided in an IP-capable MSC facilitates efficient call routing over an IP network without having to utilize the H.323 protocol, etc.

Set forth below in greater detail are a plurality of exemplary call routing scenarios for routing calls in an integrated telecommunications network such as the  
15 network described hereinabove. More specifically, the following scenarios are provided:

- (A) MS-to-MS call routing where the called MS is in its home system;
- (B) MS-to-MS call routing where the called MS is roaming;
- (C) PSTN-to-MS call routing; and
- 20 (D) MS-to-PSTN call routing.

Furthermore, each of the scenarios is provided in two exemplary embodiments, depending on certain conditions as will be described hereinbelow.

In the ensuing portions of the Detailed Description, various arrangements of an integrated telecommunications network are presented that are appropriate for the  
25 different call routing schemes, in addition to the necessary message flows wherein signaling message paths are shown in broken lines and payload-bearing paths are depicted using solid lines.

- (A) MS-TO-MS CALL ROUTING WHERE THE CALLED MS IS IN ITS HOME  
30 SYSTEM



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First, the network arrangements for two exemplary embodiments are described. The MS-to-MS call routing method is then explained in greater detail.

Referring now to FIG. 1B, depicted therein is an integrated telecommunications network 100 for effectuating a routing scheme for an MS-to-MS call in accordance with the teachings of the present invention. Three geographic regions, region1 102A, region2 102B and region3 102C, form a coverage area of the network 100. Region3 102C is not involved in the call routing scenario contemplated herein and accordingly, will not be described in this section.

A call-originating party, MS1 108A, is located in region1 102A, and is served by MSC1/VLR1 104A. An RBS1 106A provides radio access services to MS1 108A. A call-terminating party, MS2 108B, is located in region2 (home area for MS2 in this exemplary scenario), and is served by MSC2/VLR2 104B. Also, an RBS2 106B is included for providing radio access to MS2 108B.

MSC1 and MSC2 are provided with a suitable IP-IWU as described above. Also, a Location Server (LS 112) is provided within the network 100. Because MSC1/VLR1 and MSC2/VLR2 are located in two different geographic areas, the MS1-MS2 call is a long distance call. For illustrative purposes, MSC2 is treated as both the home gateway MSC and serving MSC of MS2. A signaling path 114 is provided between MSC1 and LS 112. Also, another signaling path 116 is provided between MSC1 and MSC2. An IP trunk path 118 is established therebetween for routing the voice payload associated with the call.

FIG. 1C depicts the network 100 in a form that is essentially identical to the network arrangement described above, except that the called party, that is MS2 108B, is served by an MSC (MSC3/VLR3 104) that has no direct IP-IWU interface. MSC2 operates as the destination MSC, the MSC with IP interface that is closest to the serving MSC (i.e., MSC3). Accordingly, an additional signaling path 124 and a circuit-switched trunk such as a Synchronous Transfer Mode (STM) trunk 122 (e.g., T1 or E1) are established between the destination MSC and the serving MSC. The STM trunk 122 is used in conjunction with the IP path 118 for transporting the voice payload.

FIGS. 2A and 2B depict a flow chart that describes a call routing method for

the two exemplary network arrangements set forth above. It should be appreciated that the various messages depicted in FIGS. 1B and 1C are used in setting forth the steps of the flow chart. Accordingly, FIGS. 1B and 1C may again be referred to in connection with the flow chart shown in FIGS. 2A and 2B.

5           After MSC1 receives a call initiation from MS1 including MS2's B-number (step 202), MSC1 performs a number analysis on the B-number (step 204) to determine if the call to MS2 is a long distance call (decision block 206). If it is not a long distance call, the call may be completed using conventional local call termination procedures (step 208).

10           If MSC1 determines that the call involves a different region (i.e., a long distance call), it interrogates the Location Server by sending a Service Location Protocol (SLP) message (*SERVICEREQ*), together with the B-number of the called party (step 210) to query the IP address of the destination MSC (MSC is IP-capable and its IP address is provided in the LS's database), which can also be the serving MSC (as illustrated in FIG. 1B). The destination MSC is used as a transit MSC (as  
15           illustrated in FIG. 1C) in the case where serving MSC (MSC3 in this case) is not IP-capable. The Location Server, in response, returns the IP address of MSC2, which is provided as the IP-capable MSC, via a *servicereq* message to MSC1 (steps 214 and 224). Depending on whether the serving MSC for the called MS is IP-capable or not,  
20           two scenarios emerge.

          Where MSC2 is the IP-capable serving MSC for MS2, MSC1 sends a Bearer Independent Call Control (BICC) message called *Initial Address Message (IAM)+* to MSC2, including the B-number and the IP address of MSC1 (step 216). The *IAM+* message is essentially a modified N-ISDN User Part (N-ISUP) *IAM* message, provided  
25           to effectuate the signaling as set forth herein. Upon receiving the *IAM+* message, a BICC message called *Address Complete Message (ACM)+* is sent back to MSC1 to by MSC2 (step 218). Thereafter, a BICC message called *Answer Message (ANM)+* is sent by MSC2 to MSC1 (step 220). Subsequently, the IP trunk 118 is established between MSC1 and MSC2 via Real-time Transfer Protocol (RTP) and Session  
30           Description Protocol (SDP) to convey the voice payload (step 222) associated with the call. These actions are shown in a consolidated step 222.

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Where MSC2 is only the IP-capable destination MSC because the serving MSC (MSC3) is not IP-capable, an *IAM*+ message is also initially sent from MSC1 to MSC2, after receiving the result from the LS (step 226). Subsequently, an ISUP *IAM* message is forwarded by MSC2 to MSC3 (step 228). MSC3 then sends an *ACM* message to MSC2 as an acknowledgment of the *IAM* message (step 230). Thereafter, upon receiving the *ACM* message, MSC2 initiates a BICC *ACM*+ message to MSC1 (step 232). The ISUP *ANM* message is then sent by MSC3 to MSC2 (step 234), which is forwarded by MSC2 to MSC1 by sending the BICC *ANM*+ message (step 236). The STM trunk 122 is thereby established between MSC2 and MSC3. The IP trunk 118 is subsequently established between MSC1 and MSC2 via RTP and SDP to convey the voice payload as shown in the consolidated step 238.

(B) MS-TO-MS CALL ROUTING WHERE THE CALLED MS IS ROAMING

FIGS. 3A and 3B depict the relevant network arrangements for effectuating MS-to-MS call routing where the called MS is roaming. It is clear that the network arrangements shown herein are similar to those described above. Accordingly, only the salient features of FIGS. 3A and 3B are set forth herein.

The called MS 108B is no longer located in its home area, that is, region2 102B. Rather, it is now located in region3, being served by MSC4/VLR4 104D. An RBS4 106D is included in the infrastructure of the region for providing radio access services to MS2. MSC2 is still provided as the home gateway MSC of MS2. Whereas in FIG. 3A, each of the call-originating MSC (MSC1), home gateway MSC (MSC2), and the serving MSC (MSC4) have IP interfaces, FIG. 3B depicts the scenario where the serving MSC (MSC4) does not have an IP interface and, accordingly, has to connect to a destination MSC (MSC3 in this exemplary embodiment; MSC3 is the IP-capable MSC that is geographically closest to MSC4) via an Inter-Exchange Carrier (IXC) 510B that is disposed between region2 and region3. Consequently, the network arrangement in FIG. 3B depicts STM trunks (path 170 and path 172) for the IXC connection. FIG. 3A, on the other hand, illustrates a direct IP connection 162 between the originating MSC and serving MSC for call routing.

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FIGS. 4A - 4D depict a flow chart for the call routing scenarios described above. Once again, FIGS. 3A and 3B may be referred to for locating appropriate signaling messages referenced in the flow chart. Further, it should be appreciated that most of the steps effectuated in this flow chart are similar to the steps described in the flow chart of FIGS. 2A and 2B. Accordingly, a concise description of the call routing method for each of the scenarios is set forth below without explicitly referring to the reference numerals of the flow chart shown in FIGS. 4A - 4D.

Where the serving MSC (MSC4) is provided with the IP interface, the call routing steps are as follows. After MSC1 receives a call initiation from MS1 including MS2's B-number, MSC1 performs B-number analysis to determine if the call is a long distance call. If so, MSC1 interrogates LS by sending the SLP *SERVICEREQ* message including the B-number to query the IP address of the destination MSC. Since MSC2 is provided to be the home gateway MSC of MS2, the SLP *servicereq* message (transmitted back to MSC1 by LS) includes the IP address of the home gateway MSC, i.e., MSC2. A BICC *IAM+* message is then sent by MSC1 to MSC2, including the B-number and MSC1's IP address.

Upon receiving the *IAM+* message, MSC2/VLR2 checks its record and determines that MS2 is not in its home system, i.e., MS2 has roamed out. An ANSI-41 *LOCREQ* is transmitted by MSC2 to HLR to query the location of the serving MSC that currently serves MS2. Upon receiving the *LOCREQ* message, HLR verifies the active services and queries the serving MSC4/VLR4 in region3 by transmitting an ANSI-41 *ROUTREQ* message. The pre-routing call setup is done by MSC4 by means of paging MS2. The serving MSC4/VLR4 replies with an ANSI-41 *routrreq* message containing the routing number (TLDN). Thereafter, HLR sends the answer message *locreq* including the TLDN to MSC2.

An SLP *SERVICEREQ* message including the TLDN of MSC4 is then sent by MSC2 to LS to query the IP address of MSC4. The returned *servicereq* message from LS contains the IP address of MSC4. Thereafter, a BICC *IAM+* message is sent by MSC2 to MSC4. After receiving the *IAM+* message, a BICC *ACM+* message is sent to MSC2 by MSC4 to acknowledge the *IAM+* message. A BICC *ACM+* message is then forwarded by MSC2 to MSC1 as an acknowledgment. Afterwards, a BICC

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*ANM+* message is sent to MSC2 by MSC4, which is subsequently forwarded to MSC1 by MSC2. The direct IP trunk between MSC1 and MSC4 is then established via RTP and SDP to convey the voice payload associated with the call.

Regarding the case where the serving MSC does not have an IP address, the call routing process is essentially similar to the above up to the SLP *SERVICEREQ* message sent by MSC2 to LS to query the IP address of MSC4. Since in this exemplary embodiment none of the MSCs in region3 are provided with a direct IP connection, a destination MSC (i.e., MSC3) is found in region2. Thereafter, an STM trunk is established via the IXC between MSC3 and MSC4, in addition to the IP trunk between MSC1 and MSC3, for the purpose of call routing. It should, however, be understood that in other variations of the present invention, a destination MSC may be provided within region3, thereby obviating the need for the IXC.

(C) PSTN-TO-MS CALL ROUTING

FIGS. 5A and 5B depict network arrangements for effectuating two exemplary embodiments of a PSTN-to-MS call routing scheme in accordance with the teachings of the present invention. Once again, only the essentials are set forth herein because the components of the network arrangements are similar to the components described above.

A PSTN phone 504 served by a Local Exchange (LE1) 508 is provided as the call-originating entity in region1 102A. MS1 108A is provided as the call-terminating party and, in this exemplary embodiment, is roaming out of its home system (MSC2/VLR2 104B) provided in region 2. MSC3/VLR3 104C is provided as the serving system for MS1 in the visited region, region3 102C. An IXC 510A is provided between region1 and region2 for establishing an STM trunk (first circuit-switched trunk path involving trunk segments 564 and 568) between LE1 and MSC2.

In the scenario exemplified in FIG. 5A, the serving MSC (MSC3) is provided with an IP interface (i.e., IP-IWU interface). However, in the scenario illustrated in FIG. 5B, the serving MSC (MSC4) is not IP-capable and, therefore, a separate destination MSC (MSC3) is provided in FIG. 5B. Accordingly, another STM trunk

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(second circuit-switched trunk path 582) is established between MSC3 and MSC4 for the scenario illustrated in FIG. 5B. While the destination MSC (IP-capable MSC that is closest to the serving MSC) is provided within region3 (where the serving MSC is also located), one of ordinary skill should understand that in some other exemplary  
5 embodiments of the present invention, the destination MSC may be outside the region of the serving MSC, thereby necessitating the use of another IXC.

FIGS. 6A - 6C depict a flow chart for a call routing scheme for the network arrangements set forth above. The steps provided in the flow chart are believed to be self-explanatory and, in large part, similar to the steps of the call routing methods  
10 already set forth hereinabove. Accordingly, a concise description of the flow chart is provided below.

With respect to the case where the serving MSC is provided to be IP-capable, an exemplary embodiment of the PSTN-to-MS call routing scheme is as follows. After LE1 receives a call initiation from the PSTN phone including MS1's B-number,  
15 LE1 performs a number analysis and routes the call to MS1's home gateway MSC (MSC2) via IXC by sending an *IAM* message. After receiving the call, an ANSI-41 *LOCREQ* message is sent by MSC2 to HLR to query the location of the serving MSC. Upon receiving the *LOCREQ* message, HLR verifies the active services and queries the serving MSC3/VLR3 with an ANSI-41 *ROUTREQ* message. The pre-routing call  
20 setup is done by MSC3 by means of paging MS1.

The serving MSC3 replies with an ANSI-41 *routeq* message which includes the routing number (i.e., TLDN). Upon receiving the *routeq* message, HLR sends the *locreq* message including the TLDN to MSC2. Thereafter, MSC2 sends an SLP *SERVICEREQ* message containing the TLDN of the serving MSC3 to LS in order to  
25 query the IP address of MSC3. A *servicereq* message is sent back to MSC2 from LS with MSC3's IP address. A BICC *IAM+* message is then sent by MSC2 to MSC3. In response, a BICC *ACM+* message is returned by MSC3 to MSC2 in order to acknowledge the *IAM+* message. Upon receiving the *ACM+* message, MSC2 sends an ISUP *ACM* message to LE1 via IXC to acknowledge the *IAM* message sent by LE1.  
30 Afterwards, a BICC *ANM+* message is sent to MSC2 by MSC3. MSC2 then sends an ISUP *ANM* message to LE1 via IXC. Thereafter, an STM trunk between LE1 and

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MSC2 is established, in addition to an IP trunk between MSC2 and MSC3 via RTP and SDP.

With respect to the scenario where the serving MSC does not have an IP interface, the call routing process is essentially similar up to the *servicereq* message from LS which now includes the IP address of the destination MSC (i.e., MSC3). As can be seen in FIG. 5B, additional ISUP and BICC messaging is done in order to effectuate an STM trunk between MSC3 and MSC4. The call leg between MSC2 and MSC3 is still routed over an IP trunk via RTP and SDP.

10 (D) MS-TO-PSTN CALL ROUTING

FIGS. 7A and 7B depict two network arrangements for effectuating MS-to-PSTN call routing. In FIG. 7A, an IP-capable MSC (MSC3) 104C within region3 is used as a destination MSC to route the call via the IP network to the called PSTN phone 586 (served by LE3 584). In FIG. 7B, since there are no IP-capable MSCs in region3, MSC2 in region 2 (which is the closest IP-capable MSC to LE3) is used as a destination MSC to route the call via the IP network to the called PSTN phone 586.

FIGS. 8A - 8C depict a flow chart for an exemplary embodiment of the MS-to-PSTN call routing scheme for the network arrangements set forth above. Again, only a concise account thereof is set forth below.

After MSC1 receives a call initiation from MS1 together with the PSTN phone's B-number, MSC1 performs a number analysis to determine if the call is a long distance call. If so, MSC1 interrogates LS by sending an SLP *SERVICEREQ* message including the B-number to query the IP address of the destination MSC if the call is IP-routable. Since the B-number is a PSTN number and LE has no direct IP connection, a *servicereq* message including the IP address of MSC3 (closest MSC with IP capability in region3 with respect to LE3) is sent back to MSC1 from LS. A BICC *IAM+* message is then sent by MSC1 to MSC3, including the B-number and the IP address of MSC1. Upon receiving the *IAM+* message, MSC3/VLR3 sends an *IAM* message to LE3. An *ACM* message is sent back by LE3 to MSC3 as an acknowledgment to the *IAM* message. After receiving the *ACM* message, MSC3 sends

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a BICC *ACM+* message to acknowledge the *IAM+* message. An ISUP *ANM* message is sent thereafter by LE3 to MSC3. Once MSC3 receives the *ANM* message, it sends a BICC *ANM+* message to MSC1. Subsequently, an IP trunk is established between MSC1 and MSC3 via RTP and SDP, and a circuit-switched STM trunk (first circuit-switched trunk) is established between MSC3 and LE3 for carrying the voice payload.

Where there are no IP-capable MSCs in region3 (as illustrated in FIG. 7B), the SLP *servicereq* message from LS contains the IP address of MSC2, which is the geographically closest MSC to LE3. A BICC *IAM+* message is then sent by MSC1 to MSC2 together with the PSTN phone's B-number and the IP address of MSC1. After additional ISUP and BICC messaging as shown in FIG. 7B, an STP trunk (second circuit-switched trunk) is established between MSC2 and LE3 via the IXC, in addition to the IP trunk between MSC1 and MSC2, for the purpose of call routing.

Based upon the foregoing, it should be appreciated by those of ordinary skill in the art that the present solution advantageously provides an IP-based call routing scheme for use with an integrated telecommunications network having a PSN portion that is coupled to one or more CSN portions (wireless, wireline, or both). It should be apparent that the present invention efficiently utilizes the IP "backbone" for routing long distance calls by providing a cellular infrastructure entity (i.e., a Location Server) that includes a query-able database containing mapping data between routable numbers and IP addresses of entities provided with an IWU interface. It should further be appreciated that the IP call routing provided herein does not involve the H.323 protocol. Also, because the Location Server is provided to be a cellular component that can interface with IP-capable MSCs, current cellular infrastructures may be leveraged to a greater extent in integrated telecommunications networks as the components can be retrofitted with appropriate IP interfaces etc.

Further, it is believed that the operation and construction of the present invention will be apparent from the foregoing Detailed Description. While the method and system shown and described have been characterized as being preferred, it should be readily understood that various changes and modifications could be made therein without departing from the scope of the present invention as set forth in the following claims.



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**WHAT IS CLAIMED IS:**

1. A call routing method for routing a call originated by a PSTN phone to a mobile station (MS) disposed in an integrated telecommunications network having  
5 a packet-switched network (PSN) portion and a wireless circuit-switched network (CSN) portion including a Location Server (LS) containing mapping information between routing number information and Internet Protocol (IP) address information, the method comprising the steps of:
  - receiving, in a Local Exchange (LE) disposed in a PSTN portion, a call  
10 initiation message from the PSTN phone, the call initiation message including the MS's number;
  - performing a number analysis on the MS's number by the LE;
  - routing the call by the LE to a home gateway Mobile Switching Center (MSC) of the MS via an Inter-Exchange Carrier (IXC);
  - 15 determining, in the home gateway MSC, if the MS is located in a visited area, wherein the home gateway MSC includes an Internet Protocol (IP) interface towards the PSN portion and a querying means to query the LS;
  - if so, querying, by the home gateway MSC, a Home Location Register (HLR), for location information of a serving MSC that currently serves the MS in the  
20 visited area;
  - responsive to the querying step by the home gateway MSC, interrogating the serving MSC by the HLR for a routing number with respect to the call originated by the PSTN phone;
  - responsive to the interrogating step by the HLR, returning the routing  
25 number by the serving MSC to the HLR;
  - forwarding the routing number to the home gateway MSC by the HLR;
  - thereafter, querying, by the home gateway MSC, the LS for a routable Internet Protocol (IP) address;
  - determining, in the LS, if an IP address of the serving MSC exists based  
30 on the routing number;
  - if so, returning the IP address of the serving MSC to the home gateway

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MSC by the LS;

based on the IP address of the serving MSC received from the LS,  
establishing an IP trunk between the home gateway MSC and the serving MSC;

5 routing the call originated by the PSTN phone to the serving MSC  
using a first circuit-switched path between the LE and the home gateway MSC via the  
IXC, in conjunction with the IP trunk between the home gateway MSC and the serving  
MSC, the IP trunk forming at least a segment of the PSN portion;

10 if the determining step in the LS determines that the serving MSC does  
not have an IP address based on the routing number, ascertaining an IP address of a  
destination MSC which includes an IP interface towards the PSN portion and is  
located closest to the serving MSC;

thereafter, forwarding the IP address of the destination MSC to the  
home gateway MSC;

15 based on the IP address of the destination MSC received from the LS,  
establishing an IP trunk between the home gateway MSC and the destination MSC;  
and

20 routing the call originated by the PSTN phone to the serving MSC  
using the first circuit-switched path between the LE and the home gateway MSC via  
the IXC, in conjunction with the IP trunk between the home gateway MSC and the  
destination MSC, the IP trunk forming at least a segment of the PSN portion, and a  
second circuit-switched path between the destination MSC and the serving MSC.

2. The call routing method as set forth in claim 1, wherein the step of  
querying the HLR by the home gateway MSC is effectuated by sending an ANSI-41  
25 LOCREQ message from home gateway MSC to the HLR.

3. The call routing method as set forth in claim 2, wherein the step of  
interrogating the serving MSC by the HLR is effectuated by sending an ANSI-41  
ROUTREQ message from the HLR to the serving MSC.

30

4. The call routing method as set forth in claim 3, further comprising the

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step of paging the MS by the serving MSC for a pre-routing call setup.

5. The call routing method as set forth in claim 3, wherein the step of querying the LS by the home gateway MSC is effectuated by sending a Service  
5 Location Protocol (SLP) SERVICEREQ message from the home gateway MSC to the LS.

6. The call routing method as set forth in claim 3, wherein the step of  
10 establishing an IP trunk between the home gateway MSC and the destination MSC and the step of establishing IP trunk between the home gateway MSC and the serving MSC are effectuated by sending a plurality of Bearer Independent Call Control (BICC) messages and a plurality of Integrated Services Digital Network (ISDN) User Part (ISUP) messages among at least a subset of the following nodes: the LE, the IXC, the home gateway MSC, the serving MSC and the destination MSC.

15 7. The call routing method as set forth in claim 6, wherein the IP trunk between the home gateway MSC and the destination MSC and the IP trunk between the home gateway MSC and the serving MSC are implemented using Real-time Transfer Protocol (RTP) and Session Description Protocol (SDP) to convey a  
20 voice payload associated with the call.

8. The call routing method as set forth in claim 6, wherein the  
each of the first and second circuit-switched paths comprises a Synchronous Transfer Mode (STM) trunk.

25 9. A call routing method for routing a call originated by a mobile station (MS) to a PSTN phone disposed in an integrated telecommunications network having a packet-switched network (PSN) portion and a wireless circuit-switched network (CSN) portion including a Location Server (LS) containing mapping information  
30 between called party numbers and Internet Protocol (IP) addresses, the method comprising the steps of:

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receiving, in a Mobile Switching Center (MSC) serving the MS, a call initiation message from the MS, the call initiation message including the PSTN phone's number, wherein the MSC includes an Internet Protocol (IP) interface towards the PSN portion and a querying means to query the LS;

5 performing a number analysis on the PSTN phone's number by the MSC to determine if the call is a long distance call;

if so, interrogating the LS by the MSC to obtain an IP address of a destination MSC which includes an IP interface towards the PSN portion and is located closest to a Location Exchange (LE) that serves the PSTN phone;

10 responsive to the interrogating step, receiving the IP address of the destination MSC by the MSC serving the MS;

based on the IP address of the destination MSC received from the LS, establishing an IP trunk between the MS's MSC and the destination MSC;

15 if the destination MSC and the LE are located in a single coverage area, routing the call originated by the MS to the LE using the IP trunk between the MS's MSC and the destination MSC, in conjunction with a first circuit-switched path between the LE and the destination MSC; and

otherwise, routing the call using the IP trunk between the MS's MSC and the destination MSC, in conjunction with a second circuit-switched path between  
20 the LE and the destination MSC via an Inter-Exchange Carrier (IXC).

10. The call routing method as set forth in claim 9, wherein the step of interrogating the LS by the MSC serving the MS is effectuated by sending a Service Location Protocol (SLP) SERVICEREQ message from the MSC to the LS.

25

11. The call routing method as set forth in claim 10, wherein the step of establishing an IP trunk between the MSC and the destination MSC is effectuated by sending a plurality of Bearer Independent Call Control (BICC) messages and a plurality of Integrated Services Digital Network (ISDN) User Part (ISUP) messages  
30 among at least a subset of the following nodes: the LE, the IXC, the MSC serving the MS, and the destination MSC.

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12. The call routing method as set forth in claim 11, wherein the IP trunk between the MSC serving the MS and the destination MSC is implemented using Real-time Transfer Protocol (RTP) and Session Description Protocol (SDP) to convey a voice payload associated with the call.

5

13. The call routing method as set forth in claim 12, wherein the each of the first and second circuit-switched paths comprises a Synchronous Transfer Mode (STM) trunk.

10

14. A call routing method for routing a call originated by a first mobile station (MS) and terminating at a second MS located in its home area, each of the MSs disposed in an integrated telecommunications network which includes a packet-switched network (PSN) and a circuit-switched network (CSN) portion including a Location Server (LS) containing mapping information between called party numbers and Internet Protocol (IP) addresses, the method comprising the steps of:

15

receiving, in a Mobile Switching Center (MSC) serving the first MS, a call initiation message from the first MS, the call initiation message including the second MS's number, wherein the first MS's MSC includes an Internet Protocol (IP) interface towards the PSN portion and a querying means to query the LS;

20

performing a number analysis on the second MS's number by the MSC serving the first MS to determine if the call is a long distance call;

if so, interrogating the LS by the first MS's MSC to obtain an IP address of a serving MSC that serves the second MS in its home area, the interrogating step including sending the second MS's number from the first MS's MSC to the LS;

25

determining, in the LS, if the IP address of the serving MSC exists based on the second MS's number;

if so, returning the IP address of the serving MSC to the first MS's MSC by the LS;

30

based on the IP address of the serving MSC received from the LS, establishing an IP trunk between the first MS's MSC and the serving MSC;

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routing the call originated by the first MS to the serving MSC using the IP trunk between the first MS's MSC and the serving MSC, the IP trunk forming at least a segment of the PSN portion;

5 if the determining step in the LS determines that the serving MSC does not have an IP address based on the second MS's number, ascertaining an IP address of a destination MSC which includes an IP interface towards the PSN portion and is located closest to the serving MSC;

thereafter, forwarding the IP address of the destination MSC to the MSC serving the first MS;

10 based on the IP address of the destination MSC received from the LS, establishing an IP trunk between the first MS's MSC and the destination MSC; and

routing the call originated by the first MS to the serving MSC using the IP trunk between the first MS's MSC and the destination MSC, the IP trunk forming at least a segment of the PSN portion, and a circuit-switched path between the  
15 destination MSC and the serving MSC.

15. The call routing method as set forth in claim 14, wherein the step of interrogating the LS by the MSC serving the first MS is effectuated by sending a Service Location Protocol (SLP) SERVICEREQ message from the first MS's MSC  
20 to the LS which includes the second MS's number.

16. The call routing method as set forth in claim 15, wherein the step of establishing an IP trunk between the MSC serving the first MS and the destination MSC and the step of establishing an IP trunk between the first MS's MSC and the  
25 serving MSC are effectuated by sending a plurality of Bearer Independent Call Control (BICC) messages and a plurality of Integrated Services Digital Network (ISDN) User Part (ISUP) messages among at least a subset of the following nodes: the MSC serving the first MSC, the serving MSC serving the second MS, and the destination MSC.

30 17. The call routing method as set forth in claim 16, wherein the IP trunk between the first MS's MSC and the destination MSC and the IP trunk

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between the first MS's MSC and the second MS's serving MSC are implemented using Real-time Transfer Protocol (RTP) and Session Description Protocol (SDP) to convey a voice payload associated with the call.

5           18.    The call routing method as set forth in claim 16, wherein the circuit-switched path between the destination MSC and the serving MSC comprises a Synchronous Transfer Mode (STM) trunk.

10           19.    A call routing method for routing a call originated by a first mobile station (MS) and terminating at a second MS located outside its home area, each of the MSs disposed in an integrated telecommunications network which includes a packet-switched network (PSN) and a circuit-switched network (CSN) portion including a Location Server (LS) containing mapping information between routing numbers, called party numbers and Internet Protocol (IP) addresses, the method comprising the  
15           steps of:

                  receiving, in a Mobile Switching Center (MSC) serving the first MS, a call initiation message from the first MS, the call initiation message including the second MS's number, wherein the first MS's MSC includes an Internet Protocol (IP) interface towards the PSN portion and a querying means to query the LS;

20           performing a number analysis on the second MS's number by the MSC serving the first MS to determine if the call is a long distance call;

                  if so, interrogating the LS by the first MS's MSC to obtain an IP address of a home gateway MSC that serves the second MS in its home area, the interrogating step including sending the second MS's number from the first MS's  
25           MSC to the LS;

                  determining, in the LS, the IP address of the home gateway MSC based on the second MS's number;

                  returning, by the LS, the IP address of the home gateway MSC to the MSC serving the first MS;

30           thereafter, sending the second MS's number and the IP address of the first MS's MSC from the first MS's MSC to the home gateway MSC;

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responsive to the second MS's number received from the first MS's MSC, determining by the home gateway MSC that the second MS is located in a visited area;

5        querying, by the home gateway MSC, a Home Location Register (HLR) of the second MS to obtain the second MS's location;

      responsive to the querying step by the home gateway MSC, interrogating by the HLR a serving MSC that serves the second MS in the visited area;

      responsive to the interrogating step by the HLR, returning a routing number by the serving MSC to the HLR;

10       forwarding the routing number to the home gateway MSC by the HLR; thereafter, interrogating the LS by the home gateway MSC to obtain an IP address of the serving MSC that serves the second MS in the visited area, the interrogating step including sending the routing number received from the HLR;

      determining, in the LS, if the IP address of the serving MSC exists  
15       based on the routing number;

      if so, returning the IP address of the serving MSC to the home gateway MSC by the LS;

      forwarding the IP address of the serving MSC from the home gateway MSC to the MSC serving the first MS;

20       based on the IP address of the serving MSC received from the home gateway MSC, establishing an IP trunk between the MSC serving the first MS and the serving MSC which serves the second MS in the visited area;

      routing the call originated by the first MS to the serving MSC using the IP trunk between the first MS's MSC and the serving MSC, the IP trunk forming at  
25       least a segment of the PSN portion;

      if the determining step in the LS determines that the serving MSC does not have an IP address based on the routing number, ascertaining an IP address of a destination MSC which includes an IP interface towards the PSN portion and is located closest to the serving MSC;

30       thereafter, returning the IP address of the destination MSC to the home gateway MSC by the LS;



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forwarding the IP address of the destination MSC from the home gateway MSC to the MSC serving the first MS;

based on the IP address of the destination MSC received from the home gateway MSC, establishing an IP trunk between the MSC serving the first MS and the destination MSC; and

routing the call originated by the first MS to the serving MSC using the IP trunk between the first MS's MSC and the destination MSC, the IP trunk forming at least a segment of the PSN portion, and a circuit-switched path between the destination MSC and the serving MSC.

20. The call routing method as set forth in claim 19, wherein the step of interrogating the LS by the first MS's MSC is effectuated by sending a Service Location Protocol (SLP) SERVICEREQ message from the first MS's MSC to the LS which includes the second MS's number.

21. The call routing method as set forth in claim 20, wherein the step of interrogating the LS by the home gateway MSC is effectuated by sending a Service Location Protocol (SLP) SERVICEREQ message from the home gateway MSC to the LS which includes the routing number of the serving MSC received from the HLR.

22. The call routing method as set forth in claim 21, wherein the step of establishing an IP trunk between the first MS's MSC and the destination MSC and the step of establishing an IP trunk between the first MS's MSC and the serving MSC are effectuated by sending a plurality of Bearer Independent Call Control (BICC) messages and a plurality of Integrated Services Digital Network (ISDN) User Part (ISUP) messages among at least a subset of the following nodes: the first MS's MSC, the home gateway MSC for the second MS, the serving MSC that serves the second MS in the visited area, and the destination MSC.

23. The call routing method as set forth in claim 22, wherein the IP trunk between the first MS's MSC and the destination MSC and the IP trunk

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between the first MS's MSC and the serving MSC are implemented using Real-time Transfer Protocol (RTP) and Session Description Protocol (SDP) to convey a voice payload associated with the call.

5           24.     The call routing method as set forth in claim 22, wherein the the circuit-switched path between the destination MSC and the serving MSC comprises a Synchronous Transfer Mode (STM) trunk.

10           25.     The call routing method as set forth in claim 24, wherein the the Synchronous Transfer Mode (STM) trunk involves an Inter-Exchange Carrier disposed between the serving and destination MSCs.

15           26.     An integrated telecommunications network with a wireless circuit-switched network (CSN) portion and a Voice-over-Internet Protocol (VoIP) network portion, comprising:

              a Mobile Switching Center (MSC) serving one or more mobile subscribers, the MSC having an Internet Protocol (IP)-Interworking Unit interface towards the VoIP network portion;

20               a Location Server (LS) containing mapping information between routing numbers, called party numbers and IP addresses of entities to which a call can be routed over an IP trunk from the MSC; and

              querying means in the MSC for interrogating the LS based upon at least one of a routing number and a called party number provided to the MSC, in order to obtain an IP address from the LS for effectuating the IP trunk from the MSC based on the IP address such that a call involving a mobile subscriber that is served by the MSC  
25               is routed at least in part over the IP trunk.

              27.     The integrated telecommunications network as set forth in claim 26, wherein the querying means operates by sending a Service Location Protocol (SLP) SERVICEREQ message from the MSC to the LS.  
30

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28. The integrated telecommunications network as set forth in claim 26, wherein the IP trunk is implemented using Real-time Transfer Protocol (RTP) and Session Description Protocol (SDP) to convey a voice payload associated with the call.

5           29. The integrated telecommunications network as set forth in claim 26, further includes a PSTN wherein the call is originated by a PSTN phone served by a Local Exchange disposed in the PSTN.

10           30. The integrated telecommunications network as set forth in claim 26, further includes a PSTN wherein the call is terminated to a PSTN phone served by a Local Exchange disposed in the PSTN.

15

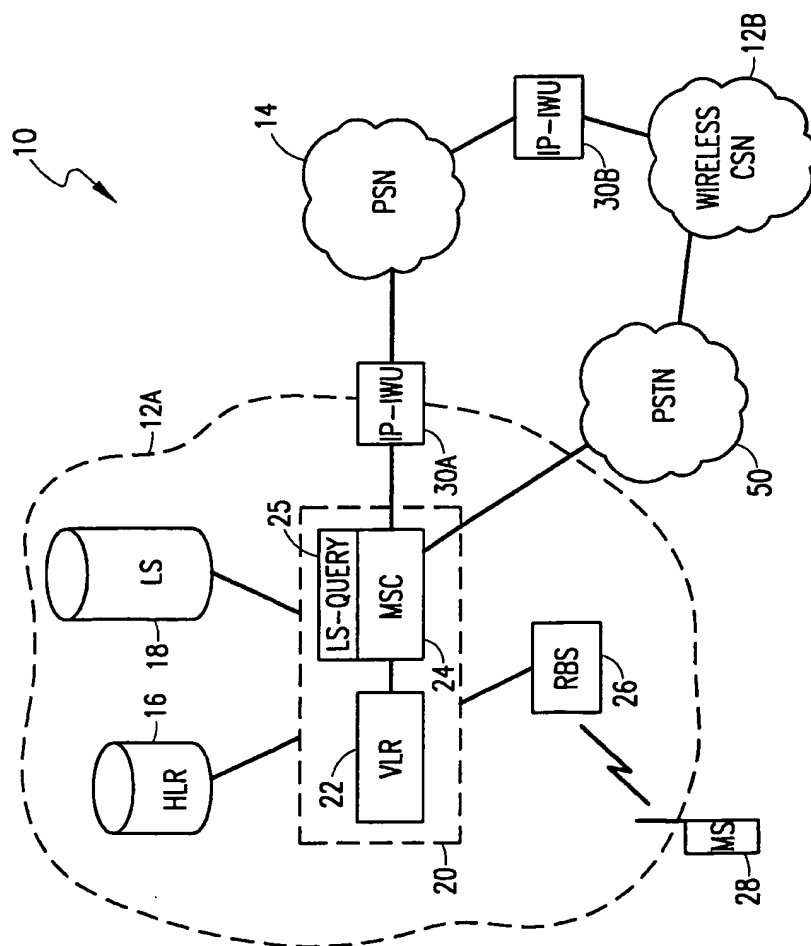


FIG. 1A

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MS-to-MS Call Case 1  
(called MS is in Home System)

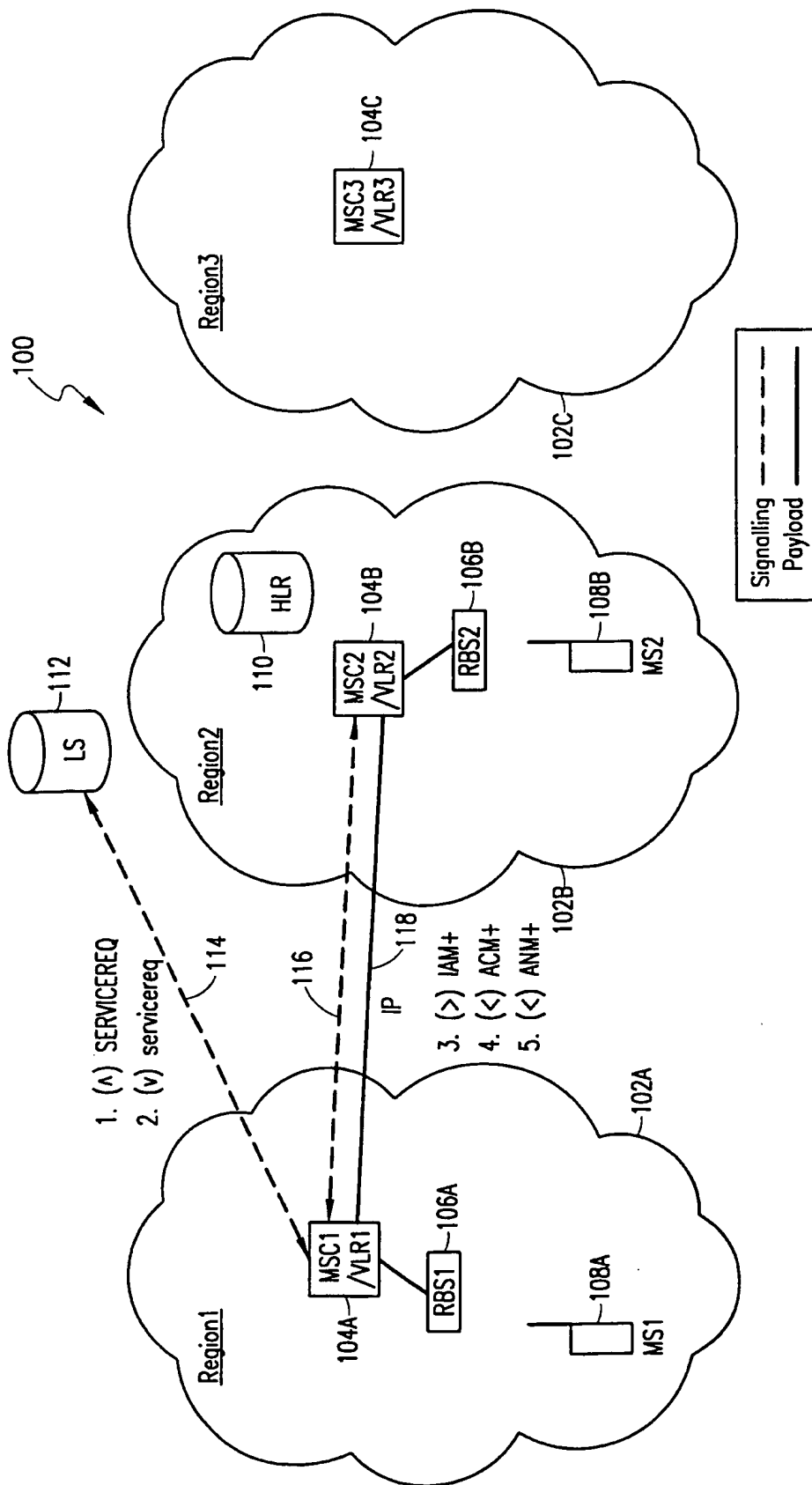
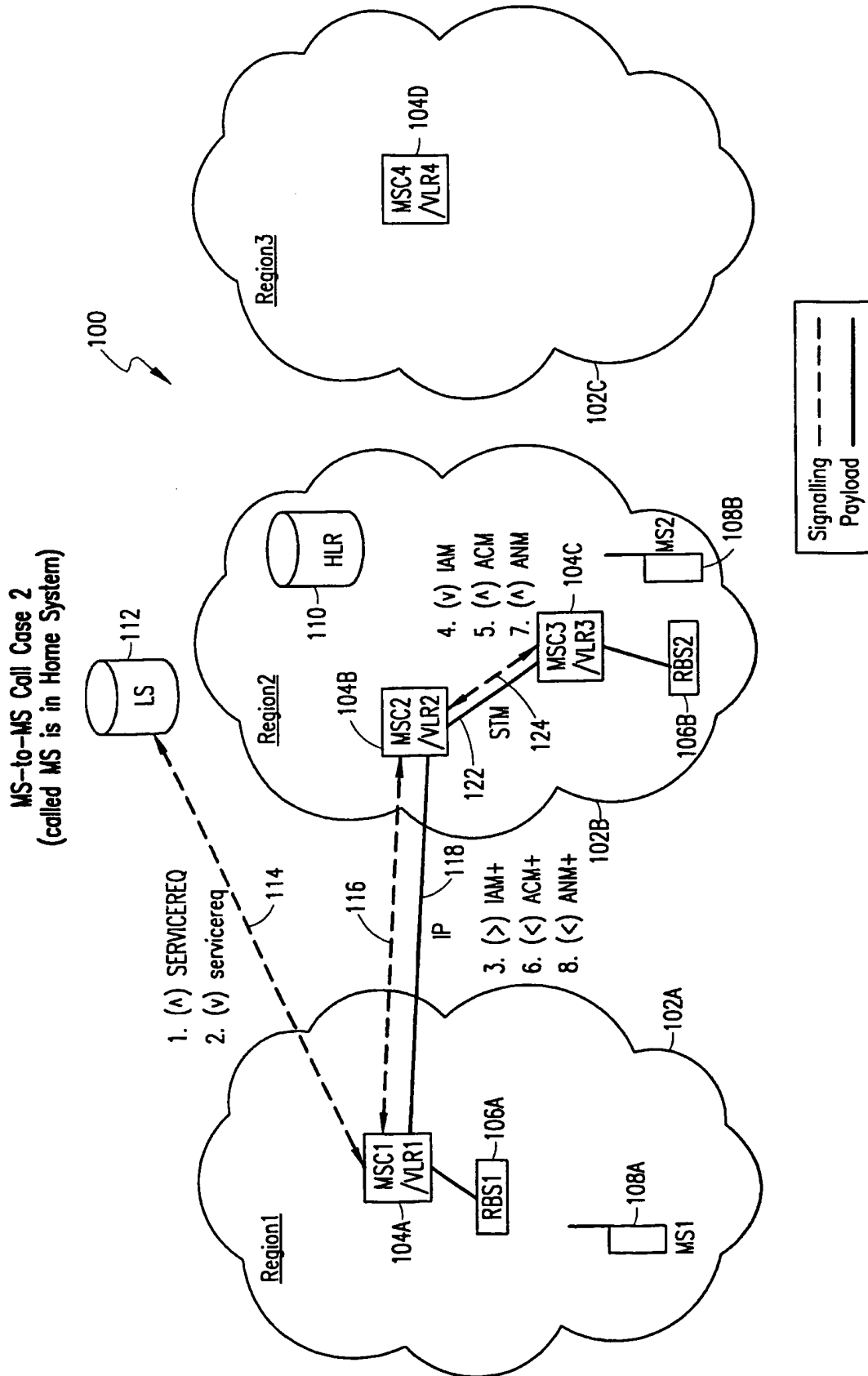
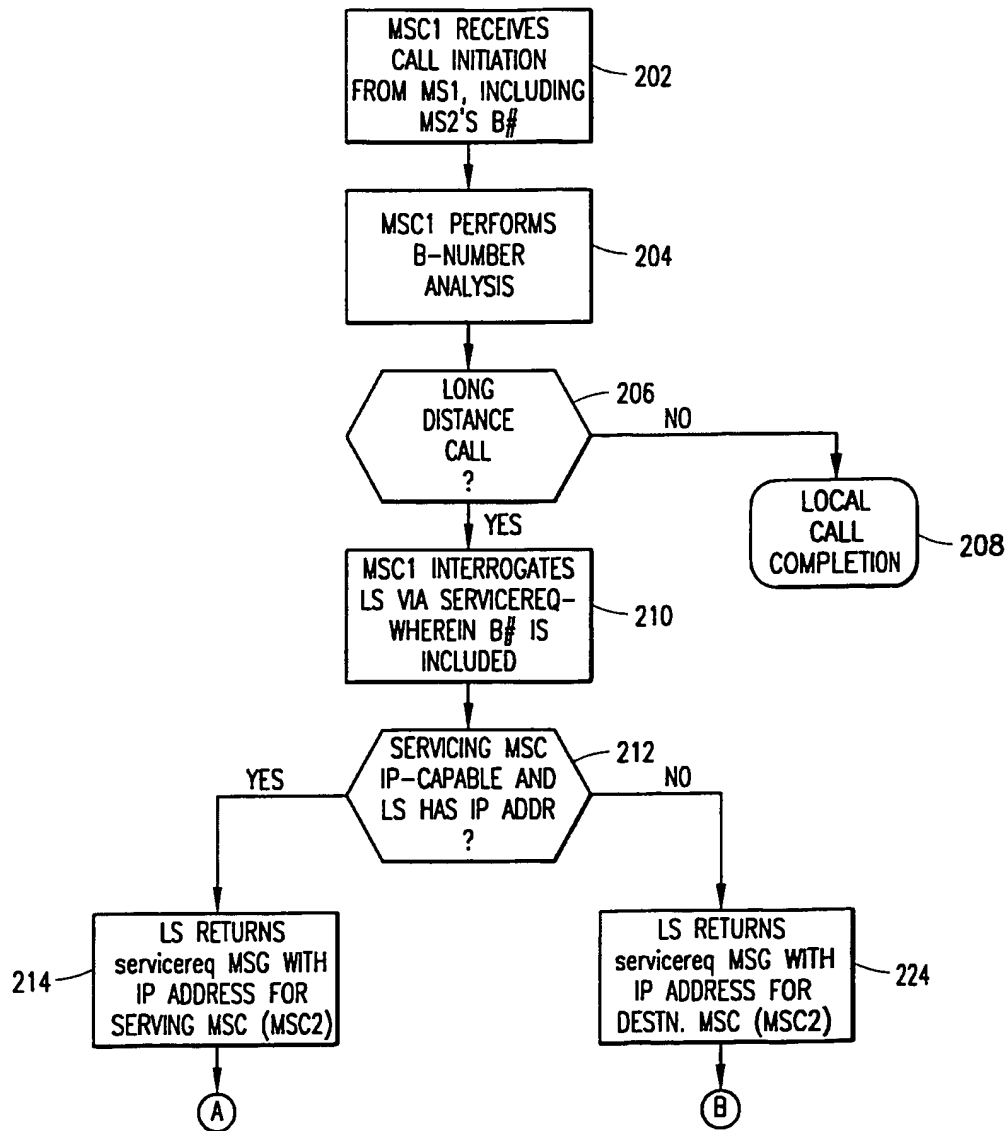


FIG. 1B



**FIG. 1C**

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TO FIG. 2B

**FIG. 2A**

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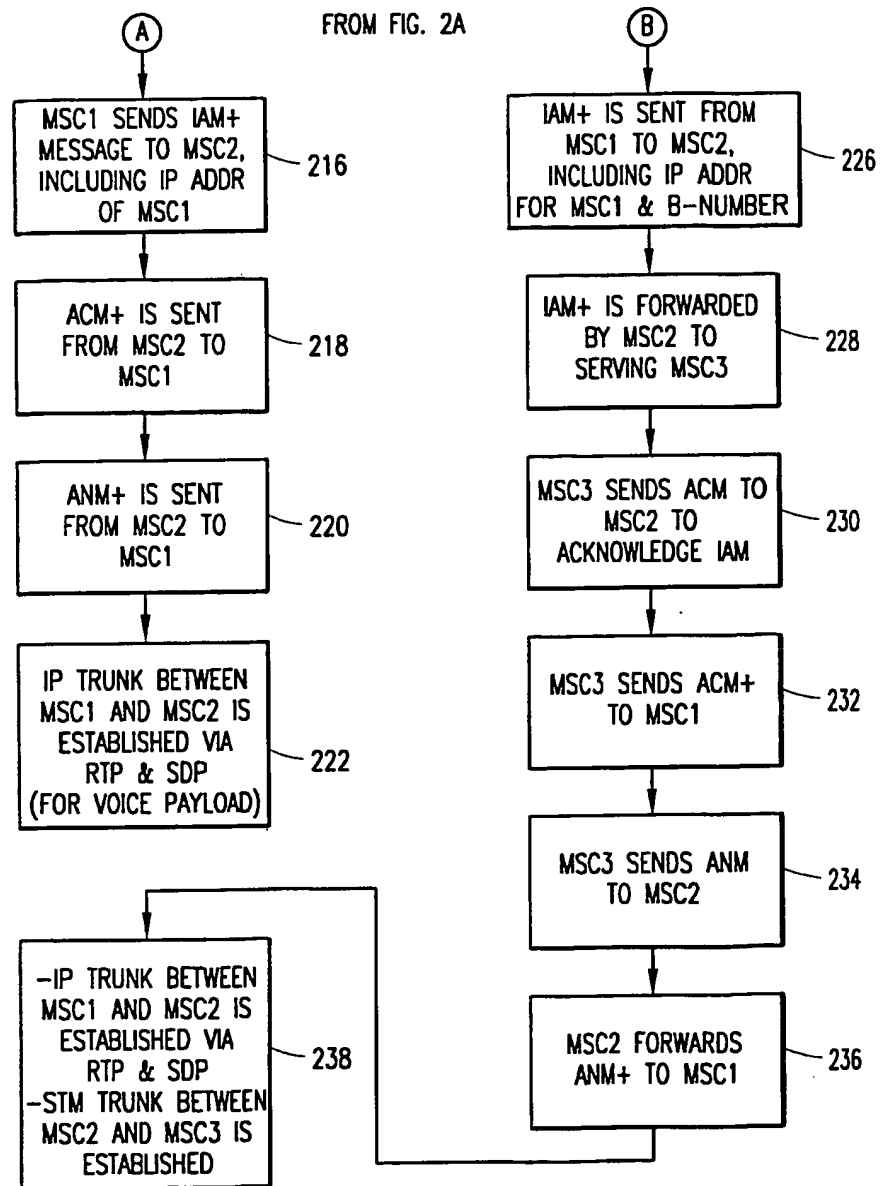


FIG. 2B



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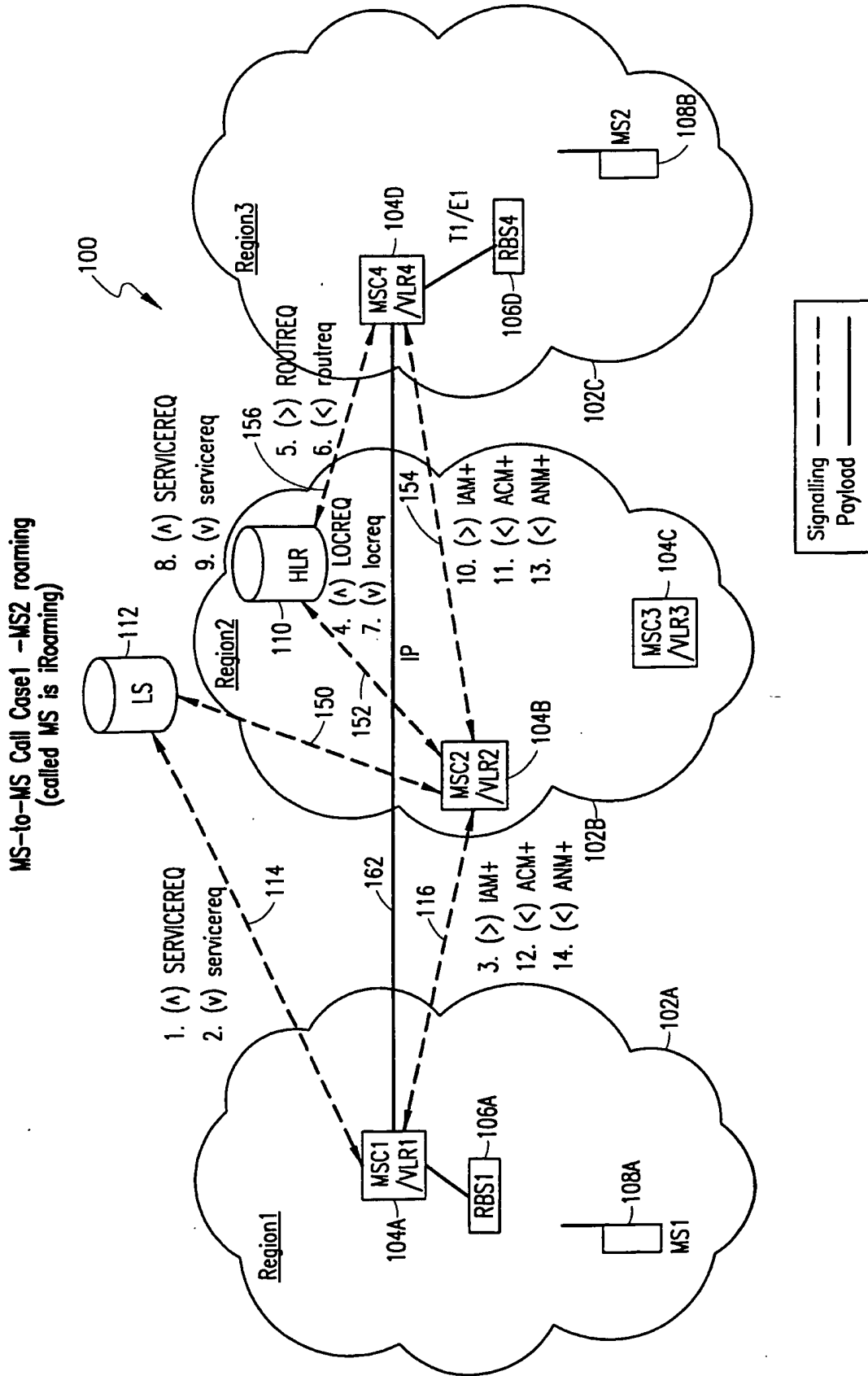
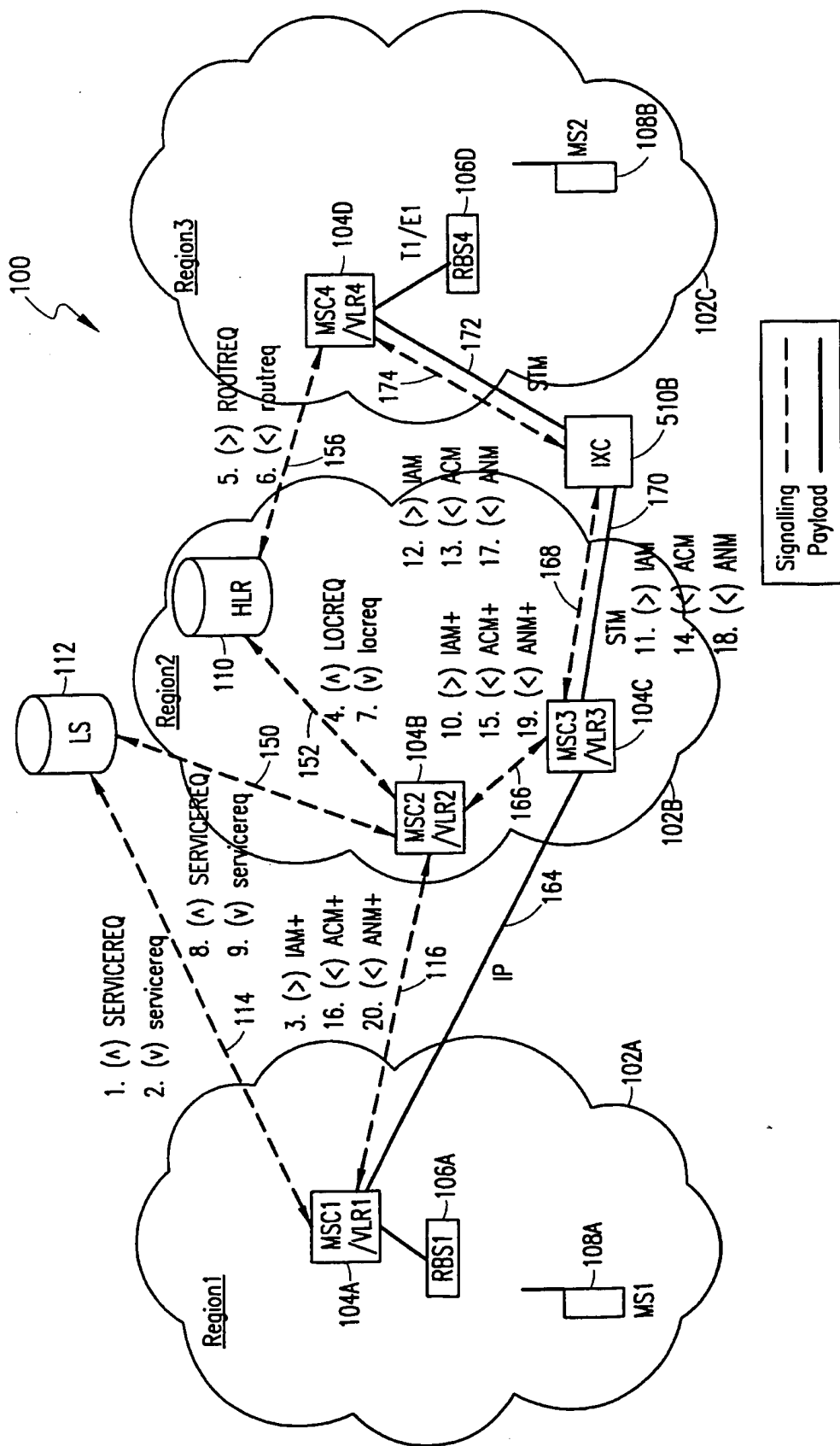


FIG. 3A

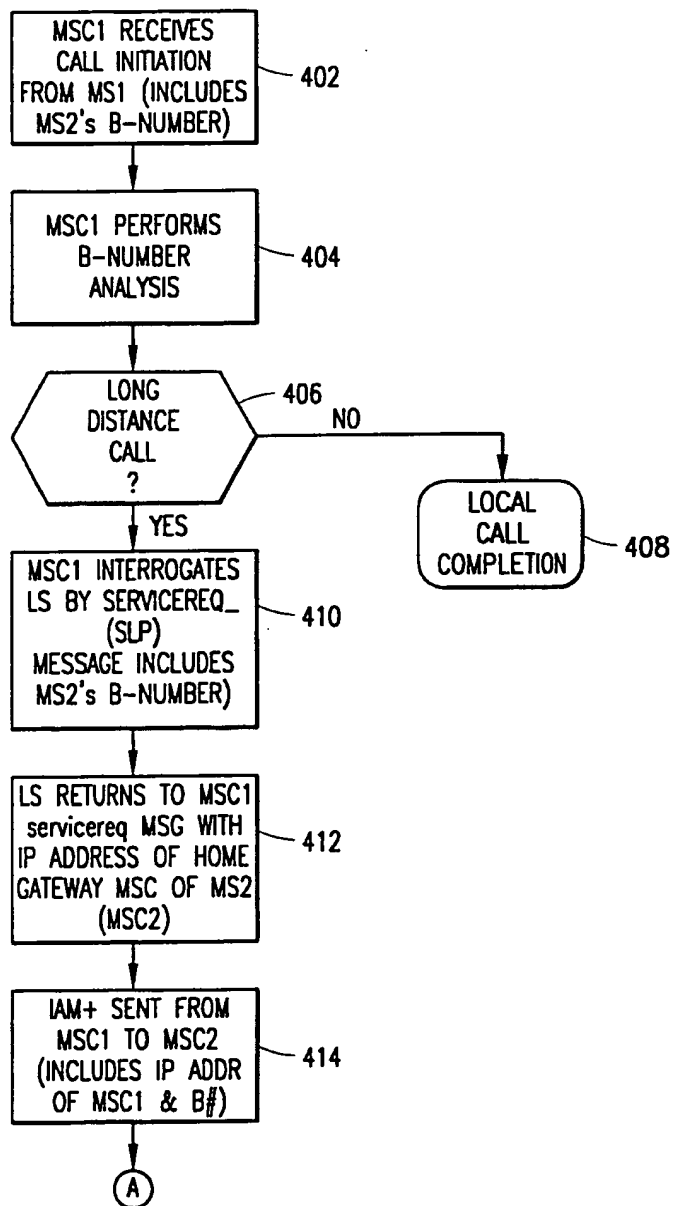
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**MS-to-MS Call Case2 -MS2 roaming  
(called MS is Roaming)**



**FIG. 3B**

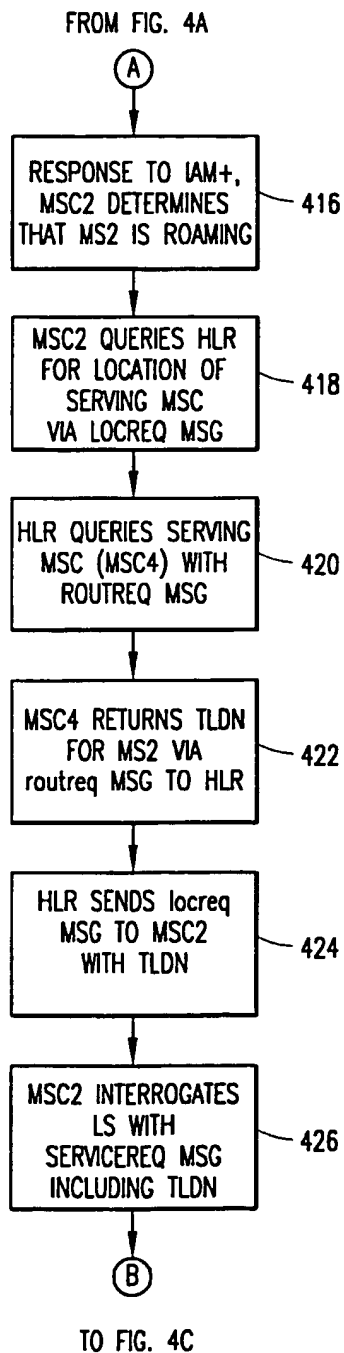
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TO FIG. 4B

**FIG. 4A**

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**FIG. 4B**

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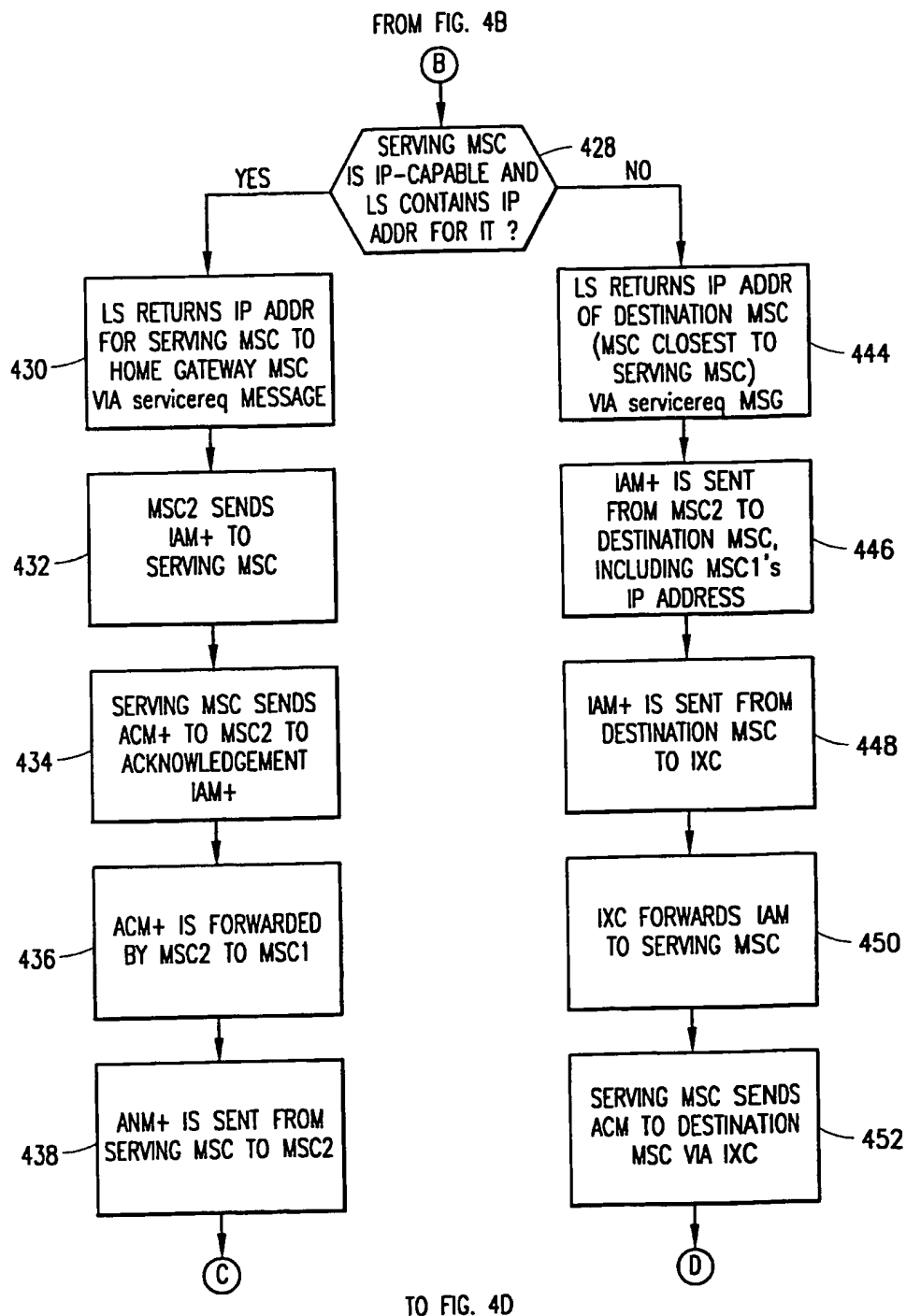


FIG. 4C

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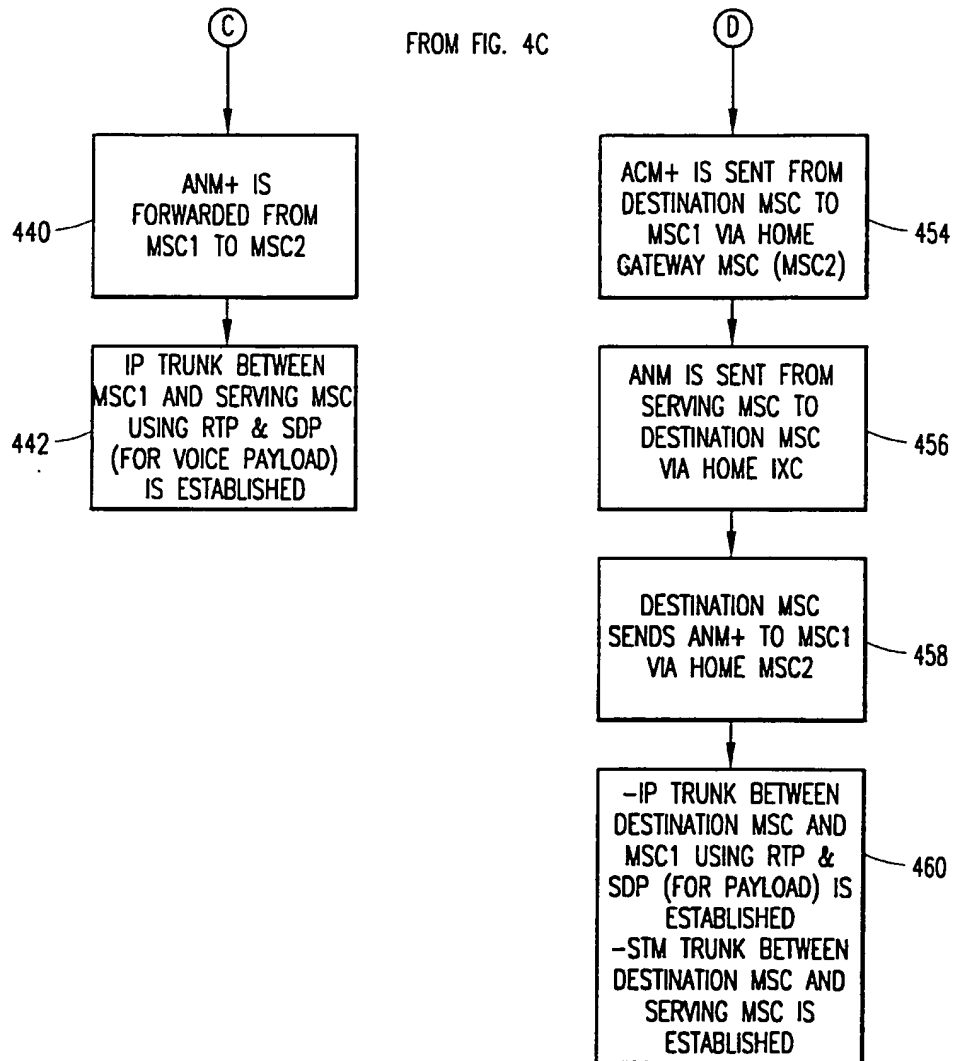


FIG. 4D

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PSTN-to-MS Call Case 1

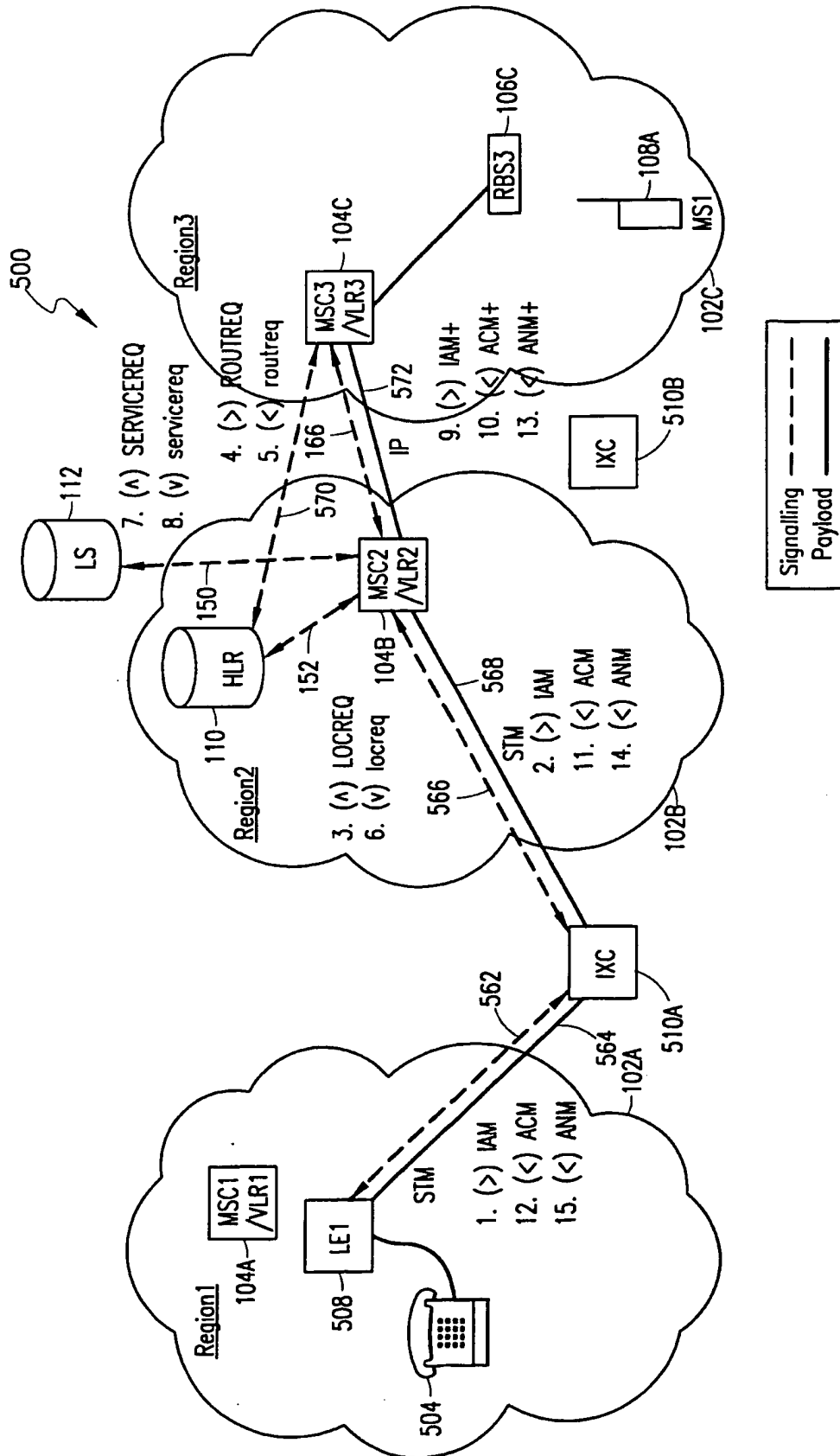


FIG. 5A

PSTN-to-MS Call Case 2

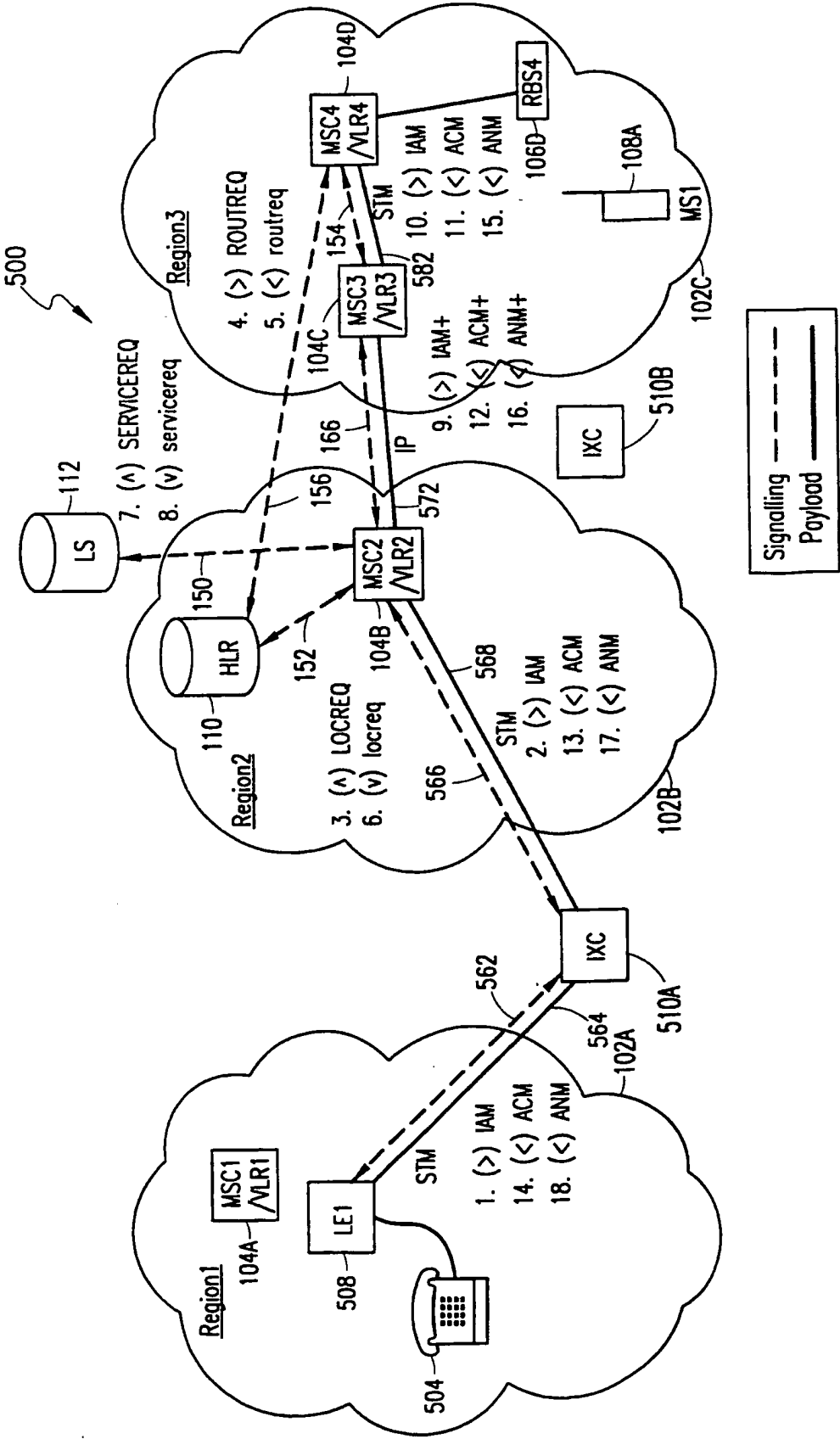
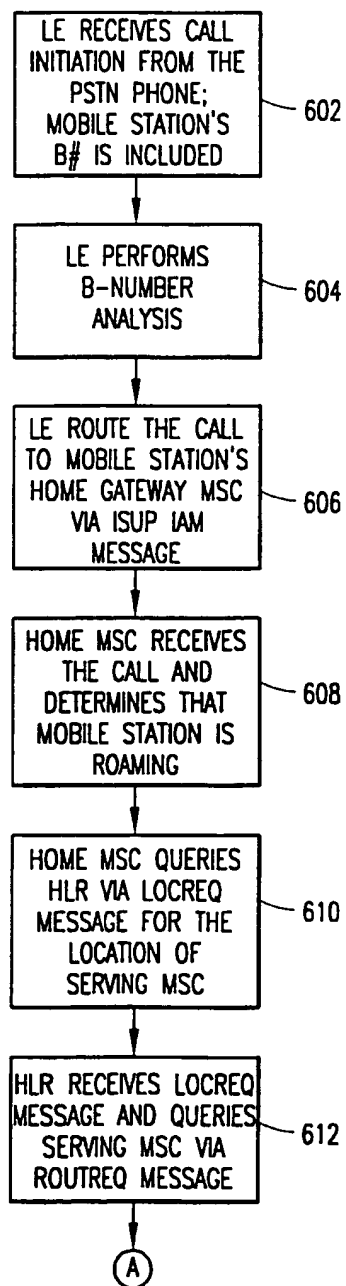


FIG. 5B



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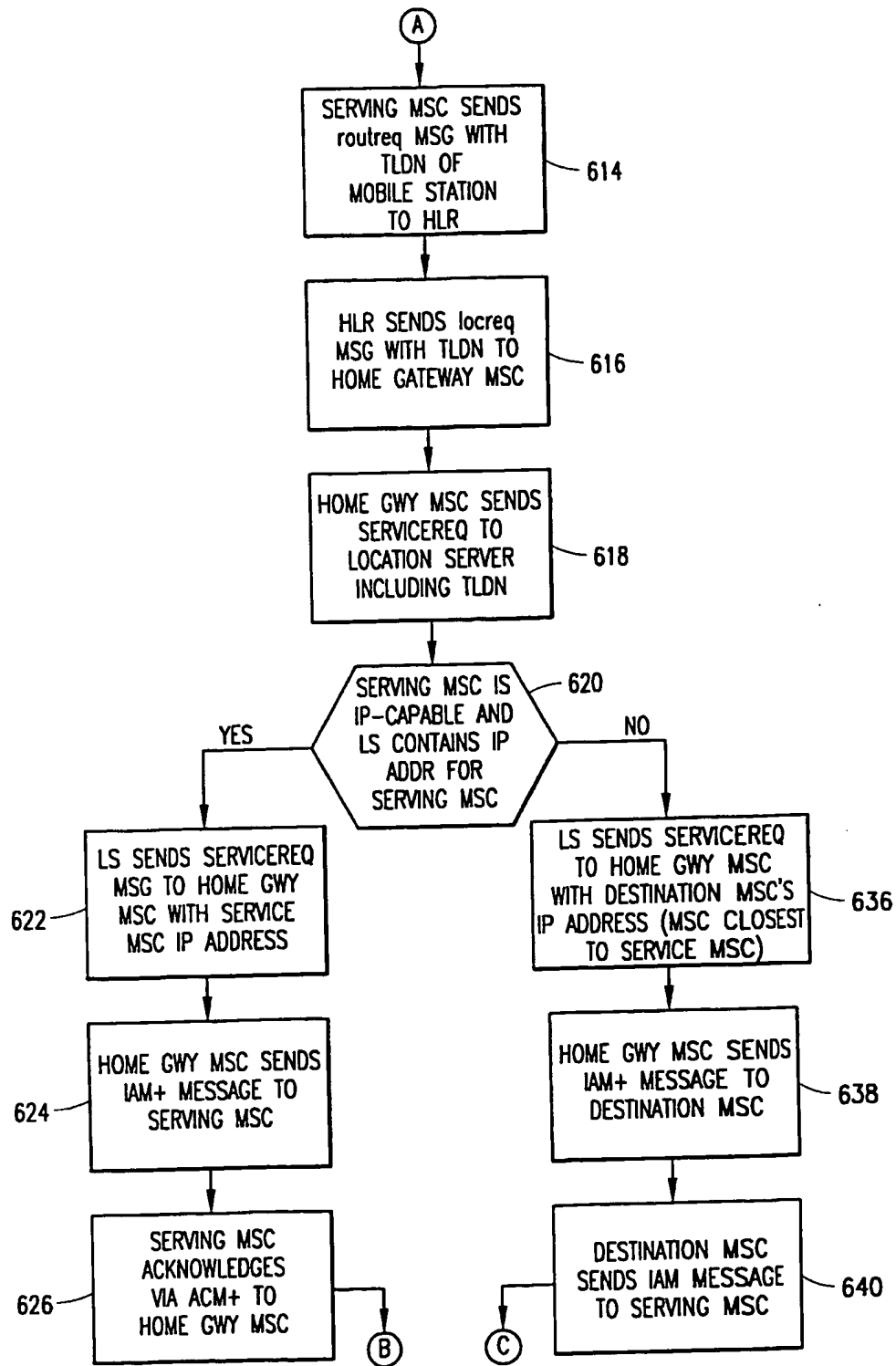


TO FIG. 6B

**FIG. 6A**

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FROM FIG. 6A



TO FIG. 6C

FIG. 6B

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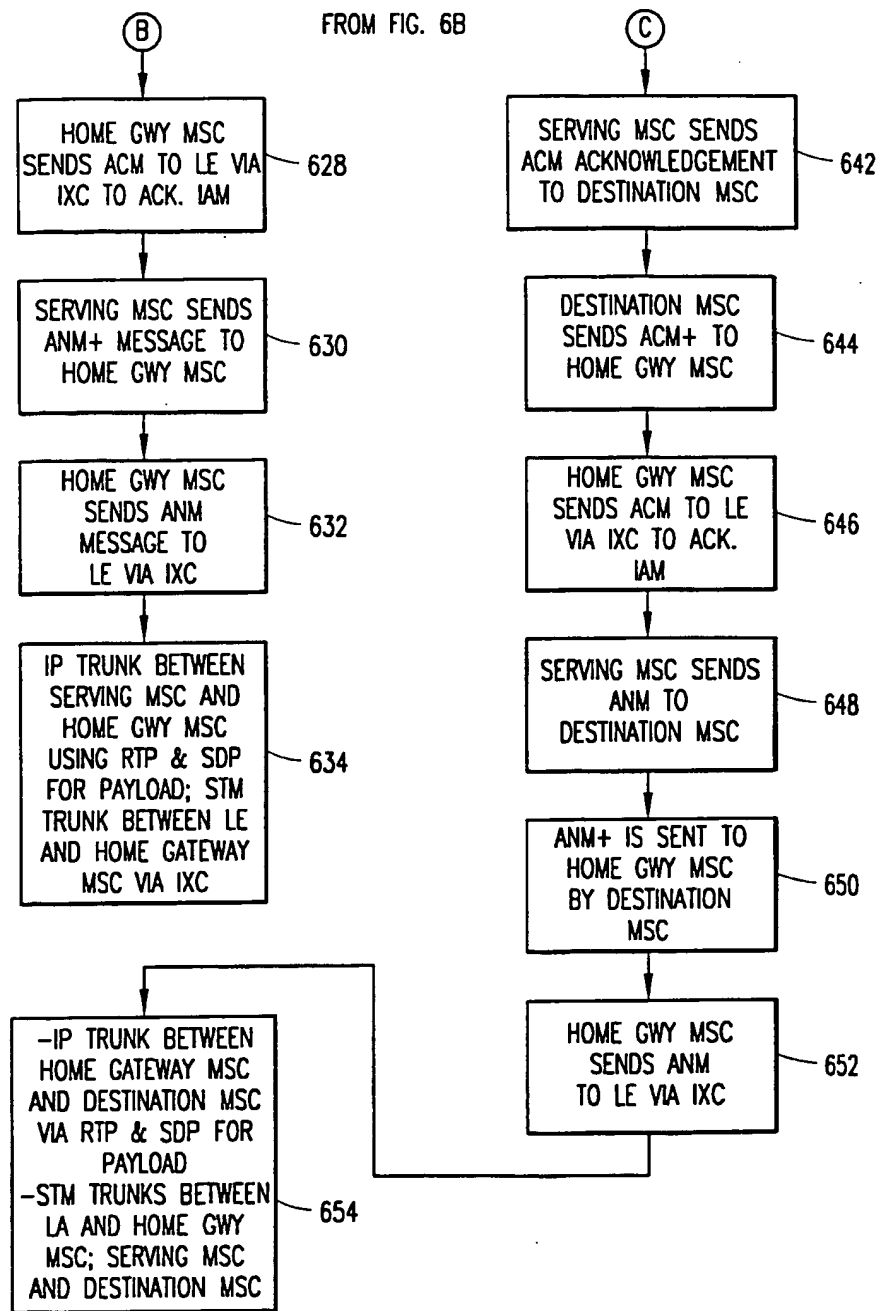


FIG. 6C

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MS-to-PSTN Call Case 1

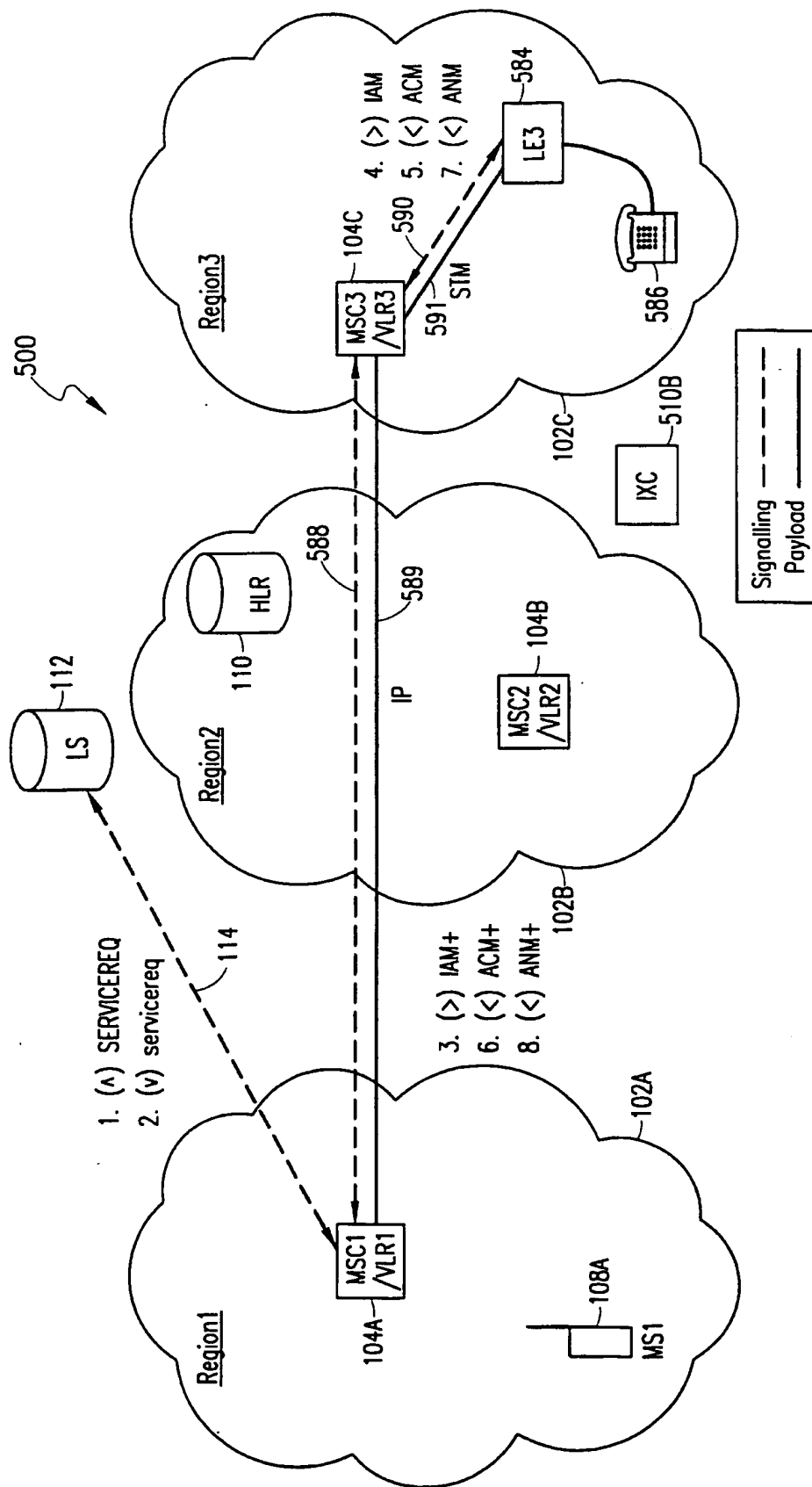


FIG. 7A

MS-to-PSTN Call Case 2

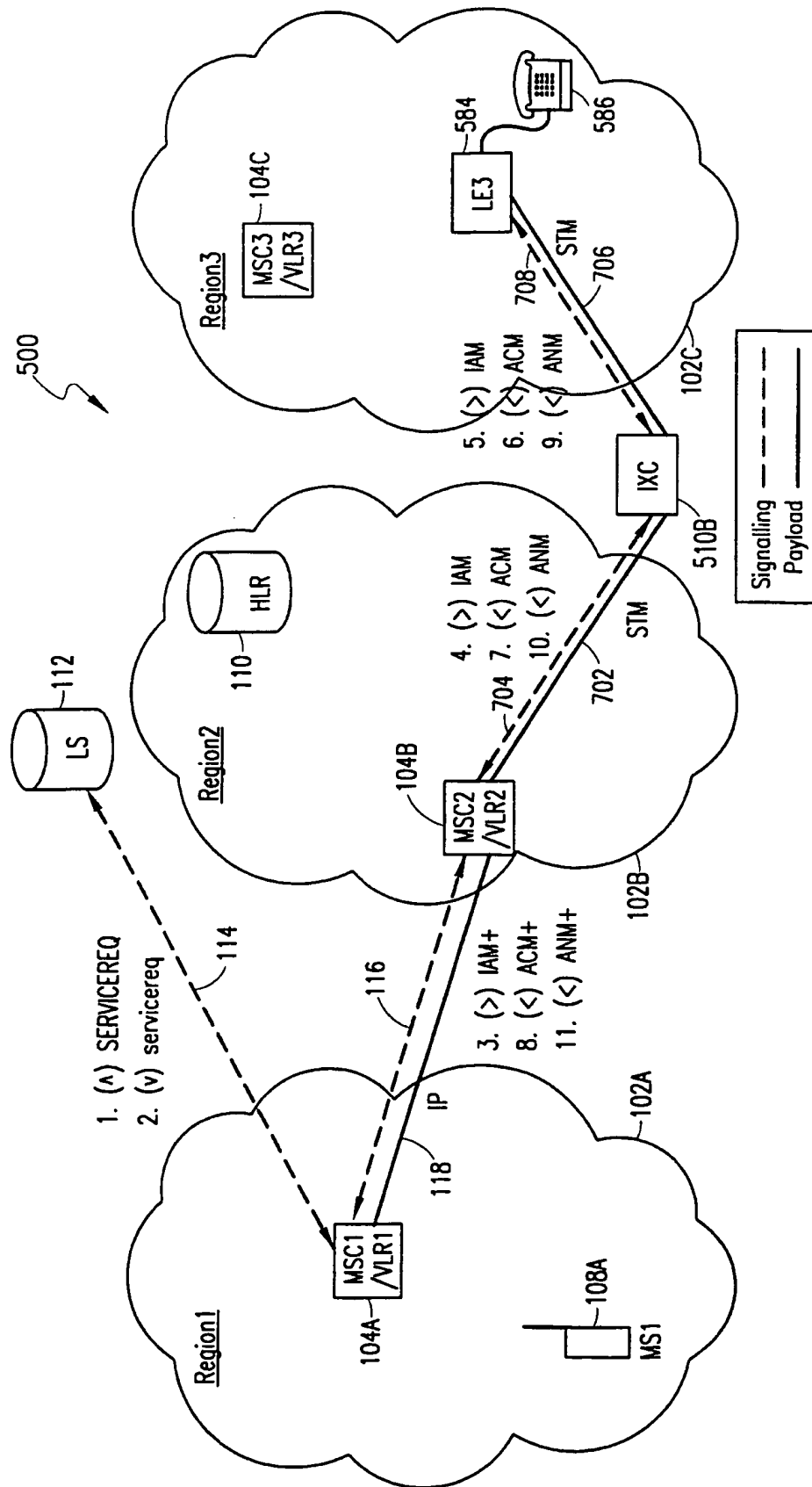


FIG. 7B

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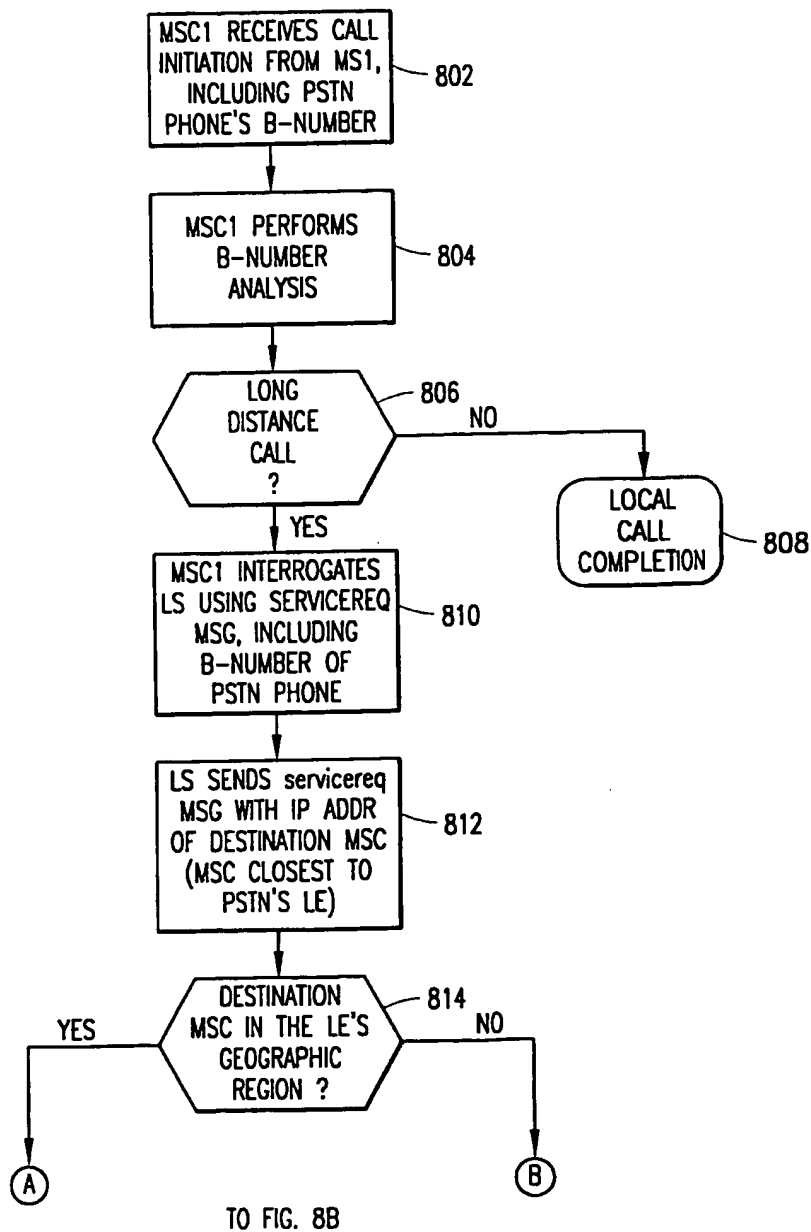
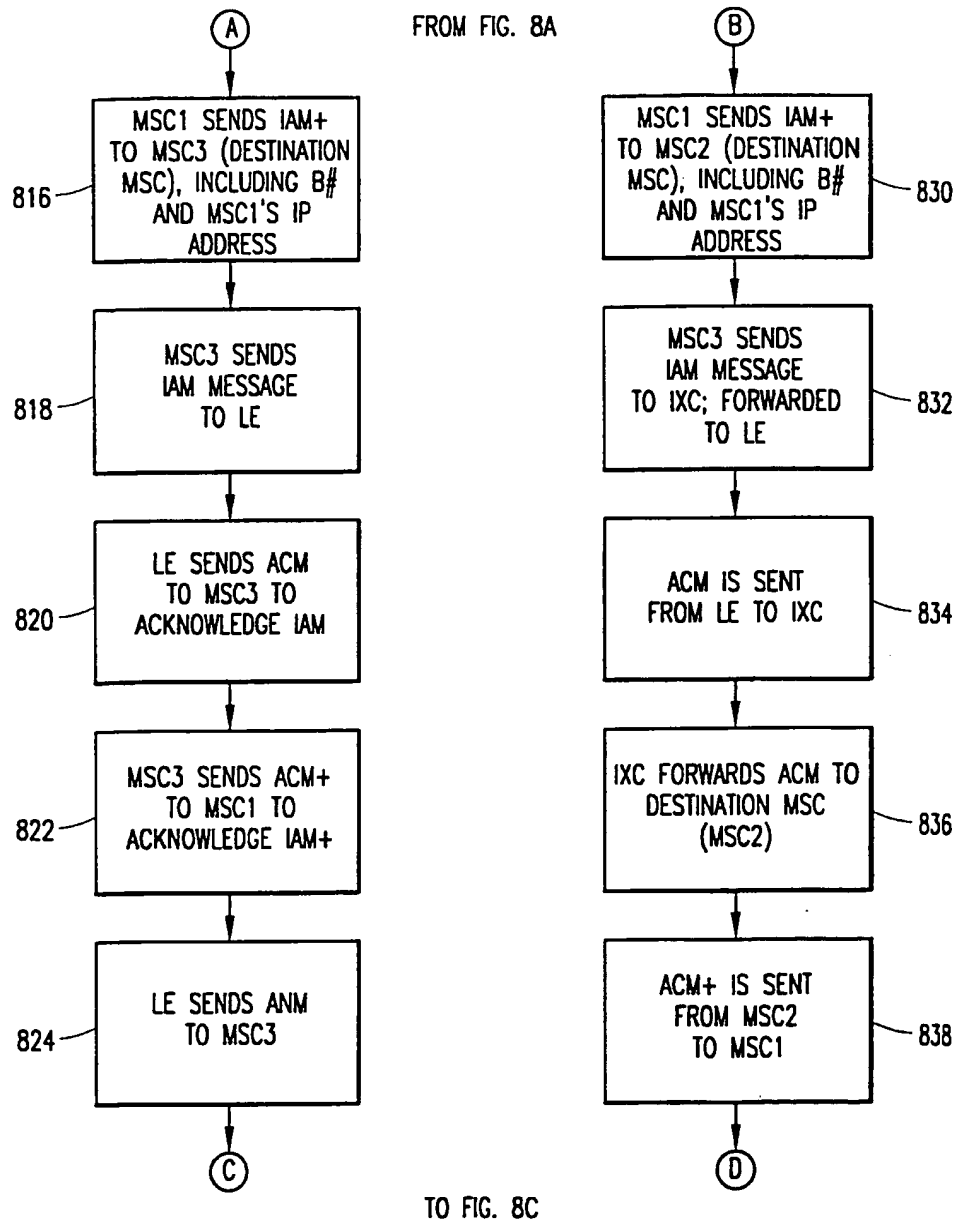
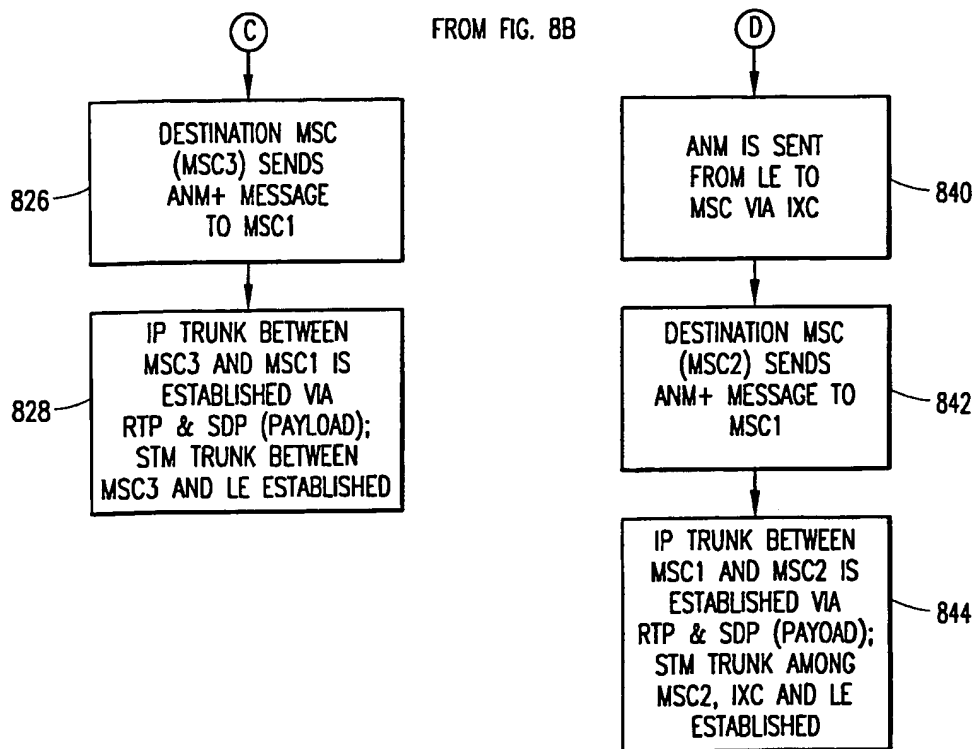


FIG. 8A

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**FIG. 8B**

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**FIG. 8C**



# INTERNATIONAL SEARCH REPORT

International application No.  
PCT/SE 00/01656

## A. CLASSIFICATION OF SUBJECT MATTER

IPC7: H04Q 7/38

According to International Patent Classification (IPC) or to both national classification and IPC

## B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC7: H04Q, H04L, H04M

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

## C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 9933250 A1 (TELEFONAKTIEBOLAGET LM ERICSSON (PUBL)), 1 July 1999 (01.07.99), page 4, line 23 - page 6, line 18; page 6, line 28 - page 7, line 20, figure 2 --	1,9,14,19,26
X	WO 9905590 A2 (STARVOX, INC.), 4 February 1999 (04.02.99), figure 2, abstract --	1,9,14,19,26
A	JP 10303990 A (LUCENT TECHNOL INC), 13 November 1998 (13.11.98) --	1-30

☒ Further documents are listed in the continuation of Box C. ☒ See patent family annex.

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Date of the actual completion of the international search

14 December 2000

Date of mailing of the international search report

05. 02. 2001

Name and mailing address of the ISA/  
European Patent Office

Authorized officer

Stefan Hansson/JAn  
Telephone No.

Facsimile No.

# INTERNATIONAL SEARCH REPORT

International application No.

PCT/SE 00/01656

## C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5781547 A (WILSON), 14 July 1998 (14.07.98), figure 2, abstract  -- -----	1-30

**INTERNATIONAL SEARCH REPORT**  
Information on patent family members

International application No.  
**PCT/SE 00/01656**

Patent document cited in search report			Publication date	Patent family member(s)		Publication date
WO	9933250	A1	01/07/99	AU	2081299 A	12/07/99
				EP	1042902 A	11/10/00
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WO	9905590	A2	04/02/99	AU	8576798 A	16/02/99
				EP	1021757 A	26/07/00
				US	6041770 A	28/03/00
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JP	10303990	A	13/11/98	NONE		
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US	5781547	A	14/07/98	NONE		
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